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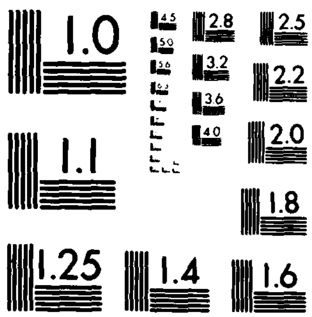

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This final report examines the implementation of a time-phased test bed for experimentation and evaluation of advanced system concepts relative to the future Defense Switched Network (DSN). After identifying issues pertinent to the DSN, a set of experiments which address these issues are developed. Experiments are ordered based on their immediacy and relative importance to DSN development. The set of experiments thus defined allows requirements for a time phased implementation of a test bed to

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be identified, and several generic test bed architectures which meet these requirements are examined. Specific architecture implementations are costed and cost/schedule profiles are generated as a function of experimental capability. The final recommended system consists of two separate test beds: a circuit switch test bed, configured around an off-the-shelf commercial switch, and directed toward the examination of nearer term and transitional issues raised by the evolving DSN; and a packet/hybrid test bed, featuring a discrete buildup of new hardware and software modules, and directed toward examination of the more advanced integrated voice and data telecommunications issues and concepts.

This report, which presents the results of the AISC/TSN study, was prepared by Dr. M. Ross, K. Garrigus, J. Gottschalck, E. Longee and L. Rinearson. Various portions of the study were performed by K. Garrigus, J. Gottschalck, Dr. E. Harrington, R. Pincus, C. Sidlo, R. Solomon and J. Ufford of GTE Systems, and Drs. W. Kelley, W. Landauer, O. Mowafi, and A. Shah of Computer Sciences Corporation, under the technical direction of Dr. M. Ross of GTE Systems. A critical in-process review of material on the Advanced Integrated System Concepts was conducted by Dr. I. Gitman and B. Occhiogrosso of DVI Communications, Inc.

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SECTION I

INTRODUCTION

In a 6 September 1979 memorandum titled "Defense Switched Network (DSN)", the Assistant Secretary of Defense - Communications, Command, Control and Intelligence (C^3I) addressed the issues of planning for future Defense Communications.[1] He noted that recent studies of CONUS AUTOVON and base telephone switching systems have highlighted the advantages of employing digital switch/transmission equipments and techniques and of planning AUTOVON switching functions at or close to the users on military bases. In addition, on-base or regional commercial satellite networks were being recommended for handling some portions of long distance traffic. The implementation of these concepts would permit a more balanced and judicious use of the existing AUTOVON, commercial and Federal Telecommunications Service (FTS) systems, by taking advantage of less costly and more efficient digital systems. Further, savings could be envisaged by satisfying most long distance switching requirements through the hub switches of the Defense Metropolitan Area Telephone Systems (DMATS) and individual base electronic switches.

As a consequence, the Defense Communications Agency (DCA) was charged with the development of a concept for the Defense Switched Network, a phased plan for implementation, and an appropriate strategy of transition to the DSN. Furthermore, the DCA, with MILDEP participation, was to refine and enlarge on concept studies currently underway which relate to specific areas of guidance outlined in the 6 September memorandum. These areas of guidance for DSN planning included: provisions for survivability for secure and non-secure high precedence users; diverse network routing of lower precedence users within a mix of media, with automatic call accounting and appropriate call control; consideratioin of AUTODIN I/II, facsimile and similar telecommunications needs; utilization of commercially available equipment; maximum sharing of DMATS hub and base switches for long distance calling; consideration of both satellite and terrestrial circuits; the maximum practical utilization of digital techniques

throughout the network; and a preference for leasing unless government owned facilities offer significant cost savings, or are required for military rotational training reasons.

In response to this direction, the DCA initiated the development of the DSN plan. As part of this activity, the DCA, with MILDEP participation, re-examined concept studies currently underway with the objective of reorienting them to be supportive of the more immediate DSN needs. One such study which was reoriented is the "Study and Evaluation of Advanced Integrated System Concepts and the Design of an Experimental Terrestrial Subnetwork (AISC/TSN)", an element of the "Experimental Integrated Switched Network (EISN)" Program.

Whereas the AISC/TSN Study had intended to address <u>advanced</u> integrated system concepts, the study was altered to address the continuum of issues and techniques from the present to the future. More specifically, the AISC/TSN objectives were redefined to be:

a. identification of key issues which are appropriate candidates for experimental examination,

b. the architecture of an experimental test bed which would cost-effectively support the examinations,

c. a plan for the phased development of the experimental test bed to support an ordered series of investigations.

In the process of identifying key nearer term and transitional issues which are appropriate candidates for examination by means of an experimental test bed, it was incumbent to evaluate for applicability at least the following issues:

a. interoperability between subscribers of different types and different networks;

b. survivability of communications via alternate routing/re-routing through diverse media and/or other networks;

c. utilization of Common Channel Signaling (CCS) for both call control and for network management;

d. Diverse network routing for low precedence calls via networks foreign to the user;

e. security capabilities with advanced integrated switching concepts;

f. provision of integrated voice/data in a common network;

g. applicability of Advanced Integrated System Concepts to the needs of DCS users in the 1990's and beyond;

h. orderly and efficient transition of the DCS from a primarily analog to a primarily digital network.

Once having identified and defined those nearer term and transitional issues which are appropriate candidates for test bed experimentation and evaluation, the AISC/TSN Study next investigated the design of a modularly expandable, flexible test bed for the test and evaluation of promising solutions to these issues. In addition, the study determined a methodology for the development of the experimental test facility to support the requirements and techniques applicable to the nearer term and transitional issues, and one also capable of being efficiently expanded to address future long term DCS issues. Among the long term considerations were: protocols and procedures strategies; precedence and priorities strategies; routing strategies; network flow control strategies; Time Assignment Speech and Data Interpolation; circuit reservation; traffic handling; and class marking.

Therefore, the purpose of this report is to present the results of the investigation of the issues raised by the DCS and the design of a flexible test bed which would accommodate evaluation of possible solutions to those issues which are appropriately supported by experimentation. The report also summarizes evaluation results and investigation of strategies for the various Advanced Integrated System Concepts.

The report comprises eight sections, including this introduction.

Section II identifies and defines the issues which are the underlying reason for the AISC/TSN program. Nearer term and transitional issues raised by the growth and evolution of the DCS, and long term issues which were the thrust of the Advanced Integrated Concepts study, are identified. Issues are grouped according to: objectives to be examined on the test bed; constraints on equipment, networks, etc.; and approaches or solutions to meeting the designated

objectives. Issues are addressed on terms of methodology for their resolution, with particular emphasis on those which are appropriate candidates for test and evaluation by means of a flexible experimental test bed facility.

Section III comprises an overview (subsections 3.2 and 3.3) of the Advanced Integrated System Concepts Study (performed during the initial period of the program) and a pragmatic treatment of nearer term concepts. The overview is divided into two parts for the purpose of presenting two different aspects of the study. Subsection 3.2 describes the Advanced Integrated System Concepts studied and evaluated, and the in-depth analyses performed to determine and justify the selection of promising approaches for implementing the future DCS. Subsection 3.3 presents a functional descriptive overview of the operation of selected candidate system concepts and a description of the strategies to be implemented in each candidate system approach.

The integrated concepts described in these two subsections address long range objectives whose realization is based on development of a new generation of switches employing advanced hybrid and packet switching techniques as well as integrated architectures. Subsection 3.3 examines integrated concepts for handling various types of traffic which make maximum use of existing equipment and techniques. These concepts focus on short range objectives, such as easing the transition to an integrated network, and are best realized through an existing family of digital switches or PABX's.

From the issues discussed in Section II, the experiments and tests to be performed on the test bed are derived. These experiments, described in Section IV, address the issues in a manner amenable to experimentation and a phased build-up of the test bed. A correlation is made between experiment groups and their associated issues. Some experiment groups impact more than one issue, e.g., routing experiments impact the issues of survivability, interoperability and cost. Relative order of experiment groups is based on the immediacy of the issues being addressed, and on the interdependencies of the

experiment groups themselves. Also identified is the case where each experiment group would be performed as soon as possible, regardless of the assigned relative order.

In Section V, the minimum test bed attributes and requirements necessary for conducting each of the experiment groups of Section IV are discussed. Requirements/characteristics are defined to permit/facilitate an incremental capability build-up to realize a flexible, modular test bed in which approaches/concepts can be efficiently implemented and experimentally evaluated.

Section VI examines various candidate test bed architectures which would support the experimental investigation of the continuum of issues - nearer term, transitional, and far term. The topics addressed are: the system operation of each architecture; the hardware and software required to conduct the different experiment types; and the advantages/disadvantages relative to nearer term/transitional or far term application.

In Section VII, estimated cost models are constructed for the candidate architectures and a projected build schedule for implementing the test bed in a time phased manner is developed. Capabilities are considered to be the tests or experiments that can be performed at each step and are consistent with the requirements identified in Section V. The rationale for creating a step or a sequence of steps is explained; the reasons for defining a step is based on a combination of experimental requirements, costs, ease of addition and engineering judgment. Interaction or interdependency of steps is identified. The basis of costing is also defined; this includes such items as commercial hardware, non-recurring engineering and high level description of new elements designed (e.g., microprocessor controlled interface or digital matrix and control).

Section VIII provides a recommendation for the experiments to be run, the test bed architecture to be implemented (with substantiative rationale), and a detailed time-phase implementation plan (with cost estimates from Section VII). An architecture based on an existing circuit switch for nearer term and transitional application, with an ability to modify software as required, and a gradual evolution to a

distributed architecture for far term application, appears to be the best approach to implementation of the test bed. This recommendation takes into consideration the relative importance of immediacy of the experiments, the time lines required in order to have an experiment impact the issues facing DCA, the cost-effectiveness of performing a particular step or implementing a feature of the test bed, and other information available to the Contractor, e.g., commercial trends. However, the recommended solution is sufficiently flexible so that a change in requirements or unforseen developments can be factored into a future decision as to which test bed implementation or series of experiments are to be followed.

<u>References - Section I</u>

1. "Defense Switched Network (DSN)", Memorandum by Gerald P. Dineen, Assistant Secretary of Defense, to various DoD departments and agencies, 6 September, 1979.

SECTION II

DEFINITION OF ISSUES

The methodology employed to identify and define those key issues, particularly nearer term and transitional DSN issues, which are appropriate candidates for test and evaluation by means of a flexible Experimental Terrestrial Subnetwork test bed facility, was based on the systematic accomplishment of a series of interrelated and mutually supporting steps. The first stage in this process was an examination of all currently available information and guidance relevant to the DSN concept. The emphasis of this examination was initially placed on the establishment of a nonpriority ordered list of concerns and requirements, especially in those cases where such items appear to be a common concern in several of the source documents and/or guidances.

The next step was to evaluate the raw list of stated, derived and deduced concerns, requirements and guidances with respect to their interrelationships; with respect to their relative urgencies; and with respect to their suitability as subjects for test and evaluation utilizing a test bed. During this evaluation, the list of concerns was stored into three categories:

Objectives: Broadly synonymous with key issues, with at least some aspects which appear as appropriate candidates for test bed examination;

Constraints: Which prevent or limit the achievement of specific objectives (and some of which apply to more than one objective);

Approaches: Techniques necessary and/or likely to permit achievement of the objectives, and/or to mitigate the limitations imposed by constraints.

The next step was to evaluate the practicability of investigating these issues/concerns by means of an experimental test bed. This was accomplished by defining desirable investigations and determining the extent to which an experimental test bed provided either the only practical means or the most apparent cost effective means of achieving the objective. The resulting unordered set of significant experiments was arranged in order of perceived "priority" as a function of the interdependencies among the candidate experiments and as a function of

the relative immediacy and importance of the issues they addressed. Consequently, the experiments derived from these issues are already ranked roughly in the required order of priority. The definitions of the individual issues nominated for experimental investigation and the rationale for the prioritized ordering follows:

Interoperability:

This issue is considered to be the highest priority of the key nearer term and transitional issues amenable to test bed experimentation. Intercommunication between subscribers of different types and different networks over multi-media networks is essential to permit the alternate routing/rerouting required for survivability of Defense Communication traffic, and to permit balanced and judicious use of the AUTOVON, commercial and FTS services, while economizing on the costs of low precedence and long distance traffic switching.

Survivability

This issue is considered to be of importance second only to interoperability. By definition, the purpose of the DCS is Defense Communications. The DCS fails in its primary purpose if telecommunications cannot be assured between secure and nonsecure high precedence (e.g., Flash and Immediate) users from the command and control, intelligence, logistics and similar communities. In addition, the DCS should permit alternate routing/rerouting through diverse media if necessary to complete a call, in the event of emergencies such as switch and/or link failure, traffic overloads, or physical destruction of part of the network.

Cost Considerations

This issue is ranked third in importance in response to the repeated emphasis placed on cost saving possibilities. Among the possibilities cited are: employment to the maximum extent practical of less costly and more efficient digital switching/ transmission systems and techniques; utilization of on-base or regional satellite terminals as well as terrestrial means to handle portions of long distance AUTOVON, commercial and FTS traffic; provisions for lower precedence calls to be handled on a diverse network routing basis within the mix of media and switches employed; employment of "automatic message accounting" to manage and control both long distance and local area calling; utilization of commercially available equipment; sharing to the maximum of hub switches and base switches for all long distance switching; a preference for leasing; and the stipulation that consideration should be given to government-owned facilities where significant cost savings can be realized.

Transition Strategies

This issue is ranked fourth in importance in recognition of the extensive existing telecommunications infrastructure, which is primarily analog in nature, utilizes primarily terrestrial means of communication, and provides various services to various user communities, but little or no intercommunication between subscribers to different networks; and the objective telecommunications system, which is to be primarily digital, utilizes satellite links extensively, provides various advanced service features and permits widespread intercommunication in more efficient and cost-effective ways. The primary issue is devising an optimum plan for the transition from the existing networks to the objective network.

System Control and Network Management

This issue is ranked fifth in importance as it depends on resolution of the interoperability and survivability issues, and is also mandated by the probable outcome of resolving these issues; however, it is also a factor of the cost considerations, and must be in consonance with the transition strategies to be developed. In the near term, the challenge will be to reconcile different and even conflicting current network management and system control procedures and requirements; during the transitional period, a coordinated system control and network management system will be required, initially for the CONUS DSN and its interaction with other networks, and eventually for the entire DSN.

Security Considerations

These considerations are ranked sixth in importance primarily due to their relatively slight amenability to experimentation via the foreseen experimental test bed facility, and not as a function of the intrinsic importance of the issue. It is clear that the DSN must meet established COMSEC objectives for at least some users from the command and control, intelligence, logistics and similar communities of users; however, investigation of means of providing such capabilities is not currently a major aim of this concept study.

Operational Capabilities

The issue of the operational capabilities to be provided to users is assigned the seventh place in order of importance of the key nearer term issues amenable to test bed experimentation. With regard to the DSN, resolution of this issue requires resolution of several of the other issues already assigned higher priorities, in particular the resolution of the interoperability, survivability and system control and network management issues, factored by resolution of the cost considerations. In addition, the designed and evolving capabilities of the hub switches and

base switches must be evaluated, as must those AUTOVON capabilities to be retained in the DSN either transitionally or permanently.

Integrated System Concepts

Test bed experimentation and evaluation of these concepts is assigned the eighth order of priority, primarily as a function of the assessment of these concepts as having more transitional and far term than nearer term applicability to DSN planning. However, experimentation regarding these advanced concepts is necessary to support resolution of such issues as interoperability and operational capabilities, and may also impact the consideration of costs.

The remainder of this section of the report reviews the guidance and requirements for each of the key issues judged to be amenable to test and evaluation via an experimental test bed facility, and summarizes for each issue specific problems which have led to the proposed experiments. The experiments themselves are described later in Section IV of this report.

2.1 INTEROPERABILITY

DSN planning will provide for the survivability of high precedence users (e.g., secure and nonsecure FLASH and IMMEDIATE precedence users) from the command and control, operations, intelligence, logistics and similar communities of users, and for handling of lower precedence calls on a diverse network routing basis within the mix of media and switches employed. One alternative for the accomplishment of the first requirement stated above involves integrating high precedence traffic fully into a DSN encompassing a multiplicity of current Defense networks (such as AUTOVON, AUTOSEVOCOM and AUTODIN) and a multiplicity of governmental and commercial networks (such as FTS and the DoD), at least during a transition phase. Another alternative would be to handle higher precedence traffic on a subnetwork within the DSN, presumably dedicated first to such higher precedence traffic.

In the near future, high usage capability will be available based on satellite and/or terrestrial linkages among hub and base systems. However, these largely independent systems have significant dissimilarities:

a.	Characteristics:	Such as their respective protocols, numbering plans, transmission plans, signaling interfaces, end equipment requirements, traffic characteristics and capacities;
b.	User provisions:	Such as routing strategies, data handling capabilities, off-network interconnections, special network features and end-to-end performance;
c.	Use constraints:	Such as Federal regulations and tariffs, security considerations and lease/buy considerations.

The issue of interoperability was analyzed as a function of its immediacy with respect to DSN planning, and as a function of the relative importance described to the issue. Based on this analysis, experiments are defined which address interoperability problems which are considered to be amenable to test bed experimentation. These experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report.

Experiments which will produce results of primary importance to the resolution of the interoperability issue include:

a. Voice Network Interfaces (Relative Order #1), which comprise the Network Interfaces Class (Class I);

b. Alternate Routing (Relative Order #2), which comprise part of the Routing Experiments Class (Class II).

Other experiments which will produce results of at least secondary importance to the resolution of the interoperability issue include:

a. Tandem Routing (Relative Order #3), which comprise part of the Routing Experiments Class (Class II);

b. CCS Formats (Relative Order #9) and CCS and Features (Relative Order #10), which together comprise the CCS Experiments Class (Class IV);

c. Combined Voice/Data (Relative Order #12) which comprise part of the Combined Voice/Data Class (Class III);

d. Data Experiments (Relative Order #15) which comprise the Data Experiments (Class VII).

2.2 SURVIVABILITY

Guidance relevant to this issue is contained within that which has been previously cited with respect to interoperability (see Section 2.1). In essence, the DCS shall provide for necessary survivability for high precedence users (e.g., secure and nonsecure FLASH and IMMEDIATE precedence users) from the command and control, operations, intelligence, logistics and similar communities of users. A further requirement is that such survivability provision be evaluated at least with respect to two alternative architectures: one in which such survivable service is furnished via full integration within the DCS; the other where it is provided via a subnetwork within the DCS.

System survivability is defined here as the capability to employ alternate routing/rerouting strategies to complete a call through diverse media, or through other governmental or commercial networks, in the event of emergencies such as switch and link failures, traffic overloads, or physical destruction of part of the network.

Introduction of the concepts of precedence handling of defense communications into this issue directs attention primarily at the AUTOVON, AUTOSEVOCOM, AUTODIN, TRI-TAC and NICS communications systems; the other candidate systems such as the FTS, ETS, and DDD do not provide such capabilities. With respect to the provision of call security, AUTOSEVOCOM, AUTODIN, TRI-TAC and NICS all make such provisions, while the ESVN and SVIP programs are directed at providing such capabilities to certain users of the AUTOVON, FTS, ETS AND DDD networks on an end-user to end-user basis, without the assistance of the switches traversed.

The problem perceived in planning for necessary survivability for high precedence users is that the networks to which such users are currently connected are typically and basically thin-line networks, (i.e., a relatively small number of trunks between switches, although, like AUTOVON, switches may be highly interconnected). If the results

of the previous effort (Interoperability; see Section 2.1) are successful, there will still remain the problems attendant on routing a call through a network providing precedence handling, but traversing networks which do not necessarily provide such service.

Similarly, the problem perceived in planning for the necessary survivability for secure and nonsecure users is again that the networks which provide such services, as distinct from those through which end-user equipments can provide call security, are typically thin line networks. Once more, a solution to the interoperability problem will still leave the problems attendant on originating a secure call from a network equipped with communications security devices and key distribution equipment and procedures, but traversing networks which do not include or provide such equipments and capabilities.

Planning for necessary survivability for secure and nonsecure high precedence users from the various communities by provision of a separate subnetwork appears likely to require a significantly higher degree of interconnection than is currently the case, and possibly a higher degree of interconnection than can be readily justified on strictly economic grounds. Planning for such survivability within an integrated DCS appears to require a solution to the interoperability problem in all of its facets.

The issue of survivability has also been analyzed as a function of its immediacy with respect to DSN planning, and as a function of the relative importance ascribed to the issue, second only to interoperability. Based on this analysis, experiments are defined which address survivability problems which are considered to be amenable to test bed experimentation. Those experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report.

Experiments which will produce results of primary importance in the resolution of the survivability issue include:

a. Alternate Routing (Relative Order #2), Tandem Routing (Relative Order #3), Packet Routing (Relative Order #5), and Flexible Trunk Experiments (Relative Order #6), which together comprise the Routing Experiments Class (Class II).

Other experiments which will produce results of at least secondary importance to the resolution of the survivability issue include:

a. Voice Network Interfaces (Relative Order #1), which comprise the Network Interfaces Class (Class I);

b. CCS Formats (Relative Order #9), and CCS and Features (Relative Order #10), which together comprise the CCS Experiments Class (Class IV);

c. Integrated Control (Relative Order #11) and Hybrid Network Control (Relative Order #13), which comprise part of the Network Control Class (Class V).

2.3 COST CONSIDERATIONS

This issue includes requirements for diverse network routing of lower precedence calls within the mix of media (including commercial and FTS means); employment of "automatic message accounting" to manage and control both long distance and local area calling; utilization of commercially available equipment; sharing to the maximum of hub switches and base switches for all long distance switching; utilization of less costly and more efficient digital switching/transmission systems; equipments and techniques to the maximum extent practical; and consideration of utilizing satellite circuits via on-base or regional satellite terminals, as well as terrestrial circuits. The desirability of optimizing utilization of transmission resources to minimize costs for lower precedence calls, to be accomplished through software control of a mix of different transmission media and networks, with the flexibility to make changes and take advantage of new traffic and procedures as appropriate, are also of concern. A key problem in this context is the transparency of foreign networks to the user.

The issues raised by cost considerations have been analyzed. Based on this analysis, experiments are defined which address problems having cost implications, and which are considered to be amenable to test bed experimentation. These experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report. Experiments which will produce results of primary importance to the consideration of costs include:

a. Tandem Routing (Relative Order #3) and Packet Routing (Relative Order #4), which comprise part of the Routing Experiments Class (Class II);

b. TASI (Time Assigned Speech Interpolation) Experiments (Relative Order #7) and Packetized Voice (Relative Order #8) which together comprise the Voice Experiments Class (Class VI).

Other experiments which will produce results of at least secondary importance to the resolution of the cost issues include:

a. Voice Network Interfaces (Relative Order #1) which comprise the Network Interfaces Class (Class I);

b. Alternate Routing (Relative Order #2) and Flexible Trunk Experiments (Relative Order #6), which comprise part of the Routing Experiments Class (Class II);

c. CCS Formats (Relative Order #9) and CCS and Features (Relative Order #10) which together comprise the CCS Experiments Class (Class IV);

d. External Monitoring (Relative Order #14) and Integrated Control (Relative Order #11) which together comprise the Network Control Class (Class V);

e. Fixed Bandwidth/Variable Boundary (Relative Order #16) and Variable Frame/Variable Boundary (Relative Order #17) experiments, which together comprise the Hybrid Experiments Class (Class VIII).

2.4 TRANSITION STRATEGIES

This issue requires refinement and enlargement on concept studies currently underway so as to reflect specific DSN guidance. The objective is to develop a well defined DSN concept for the CONUS DSN initially, to include a phased, detailed plan addressing:

* Specific hub, base switch and independent DSN features, needs, incremental sizes, etc.;
* Transmission needs for DSN and other requirements;
* System control requirements;
* Cost analysis;
* Management considerations.

In addition, this requires that the DCA in cooperation with the Services develop appropriate technical specifications for inclusion in hub and base switch procurement packages, and that the Services and Defense Agencies provide hub and base system upgrade plans to DCA for

use in designing the DSN. Finally, it requires that the DCA develop an overall DSN implementation plan to include an appropriate transition plan.

The issue of planning a strategy for transition of the DCS from a primarily all-analog to a primarily all-digital network was analyzed. Experiments are defined which address problems, some of which have implications with respect to transition strategy planning and evaluation, and which are amenable to test bed experimentation. These experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report. Experiments which will produce results of primary importance in the planning of transition strategies include:

a. Packetized Voice (Relative Order #8), which comprise part of the Voice Experiments Class (Class VI).

Other experiments which will produce results of at least secondary importance in planning transition strategies include:

a. Voice Network Interfaces (Relative Order #1), which comprise the Network Interfaces Class (Class I);

b. Alternate Routing (Relative Order #2) and Tandem Routing (Relative Order #3), which comprise part of the Routing Experiments (Class II);

c. T1 Voice/Data (Relative Order #5) and Combined Voice/Data (Relative Order #12) which together comprise the Combined Voice/Data Class (Class III);

d. CCS Formats (Relative Order #9) and CCS and Features (Relative Order #10), which together comprise the CCS Experiments Class (Class IV);

e. Integrated Control (Relative Order #11) and Hybrid Network Control (Relative Order #13), which comprise part of the Network Control Class (Class V);

f. TASI (Time Assigned Speech Interpolation) Experiments (Relative Order #7), which comprise the remainder of the Voice Experiments Class (Class VI);

g. Fixed Bandwidth/Variable Boundary (Relative Order #16) and Variable Frame/Variable Boundary (Priority Order #17), which together comprise the Hybrid Experiments Class (Class VIII);

h. Data Experiments (Priority Order #15), which comprise the Data Experiments Class (Class VII).

2.5 SYSTEM CONTROL AND NETWORK MANAGEMENT

This issue is explicit in the requirement that "automatic message accounting" be utilized to manage and control both long distance and local area calling. However, system control and/or network management requirements are also implicit in other requirements, such as:

a. Utilization of on-base or regional satellite terminals to handle portions of long distance AUTOVON, commercial and FTS traffic;

b. Diverse network routing within the mix of media (including commercial and FTS means) for lower precedence calls;

c. Consideration of AUTODIN I/II, facsimile and similar type telecommunications needs;

d. Sharing to the maximum of hub switches and base switches for all long distance switching;

e. Consideration of satellite circuits as well as terrestrial means of communications.

Though not addressing explicitly the multi-faceted system control and network management function in total, the study did address the attributes of Common Channel Signaling not only to achieve improvements in signaling capability to handle various traffic types in a single integrated network, but also as a means of transferring management information such as network monitoring data.

The related issues of System Control and Network Management were analyzed. Based on these analyses, experiments are defined which address system control and network management problems which are considered to be amenable to test bed experimentation. These experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report.

Experiments which will produce results of primary importance to the resolution of the system control and network management issue include:

a. CCS Formats (Relative Order #9) and CCS and Features (Relative Order #10), which together comprise the CCS Experiments Class (Class IV);

b. External Monitoring (Relative Order #14), Integrated Control (Relative Order #11) and Hybrid Network Control (Relative Order #13), which together comprise the Network Control Class (Class V).

Other experiments which will produce results of at least secondary importance to the resolution of the system control and network management issues include:

a. Combined Voice/Data (Relative Order #12), which comprise part of the Combined Voice and Data Class (Class III).

2.6 SECURITY CONSIDERATIONS

This issue is explicitly stated in that DSN planning shall provide for the necessary survivability for secure and nonsecure higher precedence users from the command and control, intelligence, logistics and similar communities of users. Security requirements are also implicit in the concept of diverse network routing within the mix of media and switches employed, for lower precedence calls which may originate from secure users; additionally, some of the AUTODIN I/II telecommunications to be addressed may be secure traffic.

In the nearer term Secure Voice is an example of networks and media to be integrated so as to permit intercommunication between subscribers of different types of networks over multi-media networks, and is an issue to which the applicability of a test bed should be evaluated. For the far term, advanced circuit/packet/hybrid switch concepts must be able to handle secure communications in accordance with established COMSEC objectives.

The issue of security was analyzed. Based on this analysis, experiments are defined which address problems touching on security considerations, and which are considered to be amenable to test bed experimentation. These experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report.

Due to the nature of the test bed under consideration, it is apparent that none of the experiments defined will produce results of primary importance to the resolution of security issues. However, these are some experiments which will produce results of at least secondary importance. These include:

a. Alternate Routing (Relative Order #2), Tandem Routing (Relative Order #3) and Packet Routing (Relative Order #4) which comprise part of the Routing Experiments Class (Class II);

b. TASI (Time Assigned Speech Interpolation) Experiments (Relative Order #7) and Packetized Voice (Relative Order #8), which together comprise the Voice Experiments Class (Class VI);

c. Data Experiments (Relative Order #15) which comprise the Data Experiments Class (Class VII).

2.7 OPERATIONAL CAPABILITIES

Operational capabilities may be defined as those service features, attributes and capabilities of a network and its component switches which are visible to and usable by subscribers to that network. DSN guidance does not include specific requirements with respect to such operational capabilities. However, it is clear that existing AUTOVON capabilities must continue to be made available to at least some subscribers to an integrated DSN; or at least to the subscribers to a DSN subnetwork supporting higher precedence and/or secure AUTOVON subscribers. Additional operational capabilities requirements may be inferred in that DSN planning shall address AUTODIN I/II, facsimile and similar type telecommunications needs, as well as by the concept of diverse network routing within the mix of media (including commercial and FTS means) and switches employed, for lower precedence traffic. In addition to the inferred capabilities, the study explicitly addressed operational capabilities which are integral with the interactive switching of voice and data in common network.

The issue of operational capabilities was analyzed. Based on this analysis, experiments are defined which address problems of operational capabilities which are considered to be amenable to test bed experimentation. These experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report.

Experiments which will produce results of primary importance to the resolution of the operational capabilities issue include:

a. CCS and Features (Relative Order #10), which comprise part of the CCS Experiments Class (Class IV);

b. Data Experiments (Relative Order #15) which comprise the Data Experiments Class (Class VII).

Other experiments which will produce results of at least secondary importance to the resolution of the operational capabilities issue include:

a. Voice Network Interfaces (Relative Order #1) which comprise the Network Interfaces Class (Class I).

b. Alternate Routing (Relative Order #2) and Tandem Routing (Relative Order #3), which comprise part of Routing Experiments Class (Class II);

c. T1 Voice/Data (Relative Order #5) and Combined Voice/Data (Relative Order #12) which together comprise the Combined Voice and Data Class (Class III).

2.8 INTEGRATED SYSTEM CONCEPTS

This issue is explicit in that DSN planning shall address AUTODIN I/II, facsimile and similar type telecommunications needs, as the bulk of the other requirements relate to AUTOVON and similar voice communications networks. However, there are additional implicit requirements which may be derived from the goal of arriving at a design concept that will permit a more judicious and balanced use of AUTOVON, commercial and FTS services.

The study also incorporated both explicit and implicit requirements which lead to examination of integrated system concepts as candidates for use in the DSN. Explicitly, this includes the evaluation of the following:

a. Intercommunication between subscribers of different types and networks over multimedia networks: examples of networks and media to be considered include AUTOVON, AUTODIN, DDD, FTS, Secure Voice and satellite links;

b. Integration of Voice/Data: the provision of interactive switching of voice/data in a common network;

c. Advanced concepts: advanced system concepts such as circuit/packet/hybrid switching, for future application to the needs of DCS users in the 1990's and beyond.

Implicit requirements were also derived in the study with respect to the orderly and efficient incremental enhancement of the DCS, through incorporation of advanced techniques or features, as it evolves from a primarily all-analog to a primarily all-digital system.

The issues of integrated system concepts were analyzed. Based on these analyses, experiments are defined which address problems of integrated system concepts which are considered to be amenable to test bed experimentation. These experiments are grouped as a function of their own interdependencies, and are described in Section IV of this report.

Experiments that will produce results of primary importance to the resolution of the issues of integrated system concepts include:

a. T1 Voice/Data (Relative Order #5) and Combined Voice/Data (Relative Order #12) which together comprise the Combined Voice and Data Class (Class III);

b. Fixed Bandwidth/Variable Boundary (Relative Order #16) and Variable Frame/Variable Boundary (Relative Order #17), which together comprise the Hybrid Experiments Class (Class VIII).

Other experiments which will produce results of at least secondary importance to the resolution of the issues of integrated system concepts include:

a. Flexible Trunk Experiments (Relative Order #6), which comprise part of the Routing Experiments Class (Class II);

b. Hybrid Network Control (Relative Order #13), which comprise part of the Network Control Class (Class V);

c. Packetized Voice (Relative Order #8), which comprise part of the Voice Experiments Class (Class VI.)

SECTION III

OVERVIEW OF INTEGRATED SYSTEM CONCEPTS

3.1 INTRODUCTION

This section summarizes the Advanced Integrated System Study Activity performed during the initial phases of the program as well as nearer term and transitional integrated switching concepts considered during the later phases of the program.

The remainder of this section comprises three major subsections. Subsection 3.2 summarizes the results of the Advanced Integrated System Concept Evaluation Study. Subsection 3.3 describes the architecture of several promising advanced integrated system concepts. Finally, Subsection 3.4 discusses nearer term and transitional integrated switching concepts.

3.2 ADVANCED INTEGRATED SYSTEM CONCEPT EVALUATION

This subsection describes the advanced integrated system concepts studied and evaluated, the indepth analyses performed to determine and justify the selection of promising approaches as potential contenders for implementing the future DCS, the operational features of each approach and the key features in each approach that should be experimentally evaluated to assess the relative merits (e.g., operational and complexity) of each approach.

The methodology for concept evaluation is discussed in Paragraph 3.2.1. Descriptions of each of the concepts evaluated follow in Paragraph 3.2.2. Finally, the results of the evaluation are presented in Paragraph 3.2.3.

3.2.1 Methodology

This paragraph describes the approach used for analysis and evaluation of candidate voice/data integration concepts, many of the technical assumptions that are implied or used, and the approach used during comparative evaluation of the different concepts.

Our general approach first included a search of the literature on voice/data integration. This took two forms. We thoroughly reviewed the analysis and results in the SENET-DAX Study performed under contract to the DCA in 1975-76.[1]

We reviewed similar efforts performed since that time under Independent Research and Development programs at GTE.

We also made a new search of technical literature on the topic of voice/data integration published in the last several years. We were able to find may useful approaches and results, which were incorporated into the concept evaluation study.

Our approach included development of additional analytic models to complement those obtained in the literature search. These models were computerized where necesary for parametric analysis.

The evaluation criteria called out by the SOW were systematized and unified to allow comparison of the different concepts on a common basis. Finally, the concepts were compared on this basis, and conclusions were drawn.

The remainder of this paragraph discusses the traffic basis and network considerations used in the analysis, the systematic basis for comparison of the concepts, the overall approach to evaluation, and other assumptions made for the analysis.

3.2.1.1 Models and Assumptions

3.2.1.1.1 Traffic Model - There are several categories of traffic that will be carried by an integrated network. One class consists of long messages, requiring low-delay, continuous real-time delivery, and no error control, such as voice communications. This traffic has generally been associated with circuit switching. Another type is characterized by short discrete data messages, generated asynchronously, requiring near-real-time delivery and stringent error control, such as interactive data, man/machine query/response, and message accountability processing. This type of traffic is efficiently handled by packet switching techniques. Still another class of traffic is characterized by medium to long messages of the store-and-forward type, requiring

neither continuity nor immediate delivery. These messages, typified by facsimile, narrative, and bulk data traffic, can be handled by either circuit switching or packet switching techniques.

For the purpose of analysis in this report, we assumed the following relative proportion of transmission types:

	Bits/day (x 10^9)	Percent
Voice	1,385	65.2
Data (interactive, etc.)	337	15.8
Video, fax, and bulk data	404	19.0
	2,126	

So that different integration concepts can be compared on an equitable basis, in addition to the volume data discussed above, the following packet characteristics have been assumed.

For voice packets, speech is considered sampled for various durations, just as in a real network when the sampling interval is the result of a compromise between end/end delay, link bit rate, and other system considerations. To this portion of a packet is added a constant 32-bit header[2] to arrive at the total voice packet length.

Data packets have been treated parametrically in the range of .250 to 2000 bits in 250-bit increments. The data packet header is nominally sized at 150 bits, but some sensitivity calculations were made at header lengths of 50 and 250 bits. Data packets are assumed to exist in a 10^{-5} bit error rate environment, and are retransmitted via a selective-repeat ARQ scheme.

3.2.1.1.2 Network Considerations - A simplified analysis approach was adapted. Most of the analyses were based on single link analysis assumptions. The single link results are approximate results and cannot be extended without modification to the multilink case; however, based on a high network interconnectivity assumption, some approximate extrapolation is possible.

The analytic approach used in this report would be particularly applicable to a generalized, highly interconnected, multinode,

switching network. One possible network architecture for the future DCS proposes a large number of nodes, highly interconnected through a hierarchical backbone structure, with wideband trunks used throughout the network. An example of a real military network that approaches this architecture is the AUTODIN II backbone network, in which packets travel through an average of 1.2 links to reach their destination.

3.2.1.1.3 Other Technical Assumptions - One principal factor that is implicit in concept evaluation should be spelled out. The analysis that we conducted was functional in nature, and thus was independent of implementation. We have assumed that the software and hardware necessary to perform switching and network functions can be developed and implemented in a cost-effective manner. Although it seems reasonable at this point to presume that the technology can be so developed, the usual cautions are extended to the reader.

3.2.1.2 Bases for Comparison

Criteria for the evaluation of advanced integrated system concepts (AISCs) are called out in the AISC/TSN Statement of Work. These can be systematized as:

a. Transmission efficiency, including considerations of TASI, TADI, circuit reservation, voice and data traffic handling, and effect of packet length (This criterium is particularly important because of today's high cost of bandwidth);

b. Delay of voice and data communications, including considerations of error, loss, and speech continuity;

c. Protocols and procedures, including error control and acknowledgement techniques;

d. Precedence and priority structure for voice and data communications;

e. Routing and network flow control, including recovery procedures for routing and flow control failures;

f. Service features essential to experimentation and evaluation of AISCs;

g. Classmarking for access, processing, and routing.

Evaluation criteria can be separated into three broad groups - direct quantitative measures for which explicit numeric results can be obtained; indirect quantitative measures whose effects can be less

explicitly evaluated, but which are implicit in the direct measures; and other measures where it is difficult without experimentation to evaluate network effects.

3.2.1.2.1 Direct Quantitative Measures - The measures for which analysis can produce reasonable quantitative results for integrated system concepts are the measures of transmission efficiency and delay. Transmission efficiency can itself be broken down into measures of overhead efficiency and measures of throughput. The ratio of voice and data in the network was treated parametrically in these analyses (for example, at the points of 80, 50, and 20 percent voice) in order to make a sensitivity analysis of the results.

3.2.1.2.2 Indirect Quantitative Measures - Indirect quantitative measures are those whose effects are difficult to measure exactly, but which can be measured indirectly in efficiency, throughput, and delay calculations. Such measures in this investigation are the effect of protocol approaches and the effect of routing and flow control variations.

3.2.1.2.3 Other Measures - There are several criteria that do not appear to be measurable in any significant way except by extensive simulation or experiment to evaluate the effect of variations on throughput, delay, and other direct system measures. Those criteria are: the precedence/priority structure; speech continuity, error, and loss; service features; and classmarking.

3.2.1.3 Approach to Comparative Evaluation of Concepts

Following the detailed analyses of the various concepts in the light of the specified criteria, performance results are compared. The most important bases for comparison, as might be expected, are the direct quantitative measures of transmission efficiency, throughput, and delay. Transmission efficiencies are compared over a range of voice/data ratios and with consideration of voice bit rate. The absolute values of delays are compared for the various concepts, with consideration given to the resulting throughput, a system function closely coupled to delay.

Those measures that are indirectly quantitative - protocol, routing, and flow control - are then comparatively evaluated, as are the remaining, more-qualitative measures. In these instances, where numeric results have less meaning or are simply not available, we have applied engineering judgment during the comparison.

3.2.2 Description of Concepts

This paragraph describes the candidate voice/data integration concepts which were analyzed and evaluated using the methodology described in 3.2.1.

3.2.2.1 Circuit Switching

Circuit switching dominates today's telecommunication facilities. In this technique, an entire circuit, as in analog communication, or a predetermined portion of a channel, as in time-division-multiplexed digital communication, is dedicated to continuous use of the voice or data subscribers for the duration of the call.

3.2.2.1.1 Traditional Circuit Switching - Examples of traditional circuit switching are the public telephone system and telex (switched teleprinter) system. A person or a terminal places a call by entering into the switch the directory number (address) of a person or terminal to be called. The switch then sets up a dedicated connection between suscribers consisting of a sequence of point-to-point circuits joined together by switches at the junctions between them. This connection is established by a signaling message (see Figure 3.2-1) that passes through the network. A return signal tells the source that voice or data transmission can begin. The connection exists for the two communicating parties until they decide to "hang up," or terminate the connection. During this time, they use it exclusively. Although some multiplexing can take place in a portion of the transmission system (e.g., PCM trunks), the parties will not notice it.

In the past, conventional circuit switched voice networks used analog signaling techniques such as "in-band" signaling (e.g., using analog Dual Tone Multifrequency (DTMF) or Multifrequency (MF) tones). Using the conventional analog "in-band" signaling approach, the routing signals were transmitted on the same transmission trunks used

by the voice conversation traffic; however, at present, circuit switched networks are looking toward the transmission of signaling information on high-capacity digital channels reserved solely for signaling. The high-capacity digital nature and lack of contention lead to the rapid transfer of routing information and call setup. The Bell System's Common Channel Interoffice Signaling (CCIS) is a good example of this, and will be used for analyzing the performance of the traditional circuit switching concept.

The model used for signaling traffic assumes dedicated link capacities for signaling. Figure 3.2-2 illustrates the overlay of the signaling network upon the "call-carrying" network, known as Associated Node Signaling. An associated node signaling system is used for the performance analysis of this concept.

The call-carrying network modeled consists of full-duplex digital circuits at a bit rate equal to the voice digitization rate. When a circuit has been setup, an end-to-end circuit is dedicated to subscribers and the switching nodes are initialized to transfer information from the incoming link to the outgoing link. The detailed operation of a circuit-switched network depends on the switching and transmission technology implementations. In contrast to the packet switched mode; however, the delay of circuit switched information (during the communication period) when traversing a switch is virtually constant and independent of switch load.

3.2.2.1.2 <u>Fast-Circuit Switching</u> - Fast-circuit switching causes a circuit to be established for every message that is to be sent and then disconnected after transmission. It takes advantage of the low duty cycle of interactive users by not dedicating the circuit to the user during this "think-time". The recovered capacity could be used to transfer more data, thus improving network efficiency. In addition, the channel capacity not used during circuit setup and disconnection is also available. Advanced digital switches enabling the setup or disconnection of a call circuit in 140 ms or less is assumed. Note that this assumes that no satellite links are in the network. It is likely that future switches and networks will have this capability, although none exist today. This would satisfy the

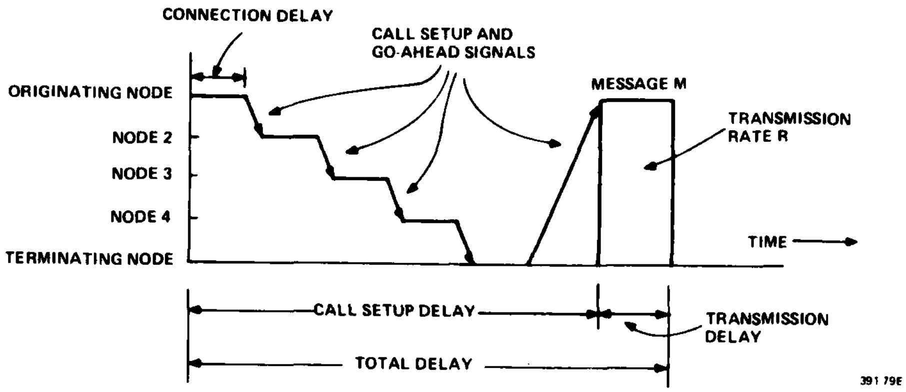


Figure 3.2-1. Circuit-Switching Network Delay

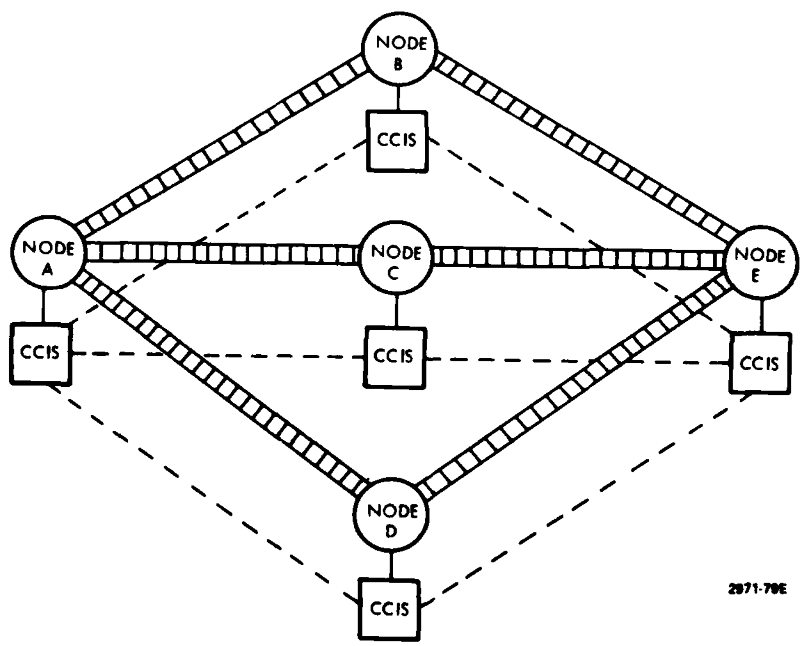


Figure 3.2-2. Circuit-Switched Network - Associated Common Channel Signaling (CCIS)

strict end-to-end delay requirements of 200 to 250 ms for interactive data applications. Only a small amount of transmission capacity would be wasted during circuit setup and disconnection because of the increased speeds involved. Recovered capacity would again help to improve transmission efficiency.

If the think-time, (i.e., the gaps between interactive messages) decreases, the efficiency of dedicating the circuits to individual users likewise increases.[3] A point is reached when the think-time is 1 to 2 seconds or less where traditional circuit switching is more cost-effective than fast-circuit switching. For call setup times greater than 280 ms but less than 1 to 2 seconds, traditional circuit switching is more cost-effective than fast-circuit switching. This is a result of (projected) switching cost being significantly higher for fast-circuit switching than for traditional circuit switching, thereby somewhat offsetting the advantage obtained by more efficient use of the transmission capacity.

The use of fast-circuit switching would thus satisfy the requirements for voice and data traffic delay and efficiency within the circuit switching structure. Typically, voice and bulk data applications would use the traditional circuit switching method, whereas interactive data would be fast-circuit switched. For interactive data, a circuit would be established for every message, held for the duration of the transaction, and then disconnected. The strict end-to-end delay requirement of 200 to 250 ms for interactive data applications would be satisfied using the fast-circuit switching concept.

3.2.2.1.3 Enhanced Circuit Switching - Enhanced circuit switching is a design concept that attempts to overcome the poor transmission efficiency associated with the traditional circuit switching of voice and interactive data (see Section 3.2.2.1.1). This can be accomplished by using traditional circuit switches supplemented with Time Assignment Speech Interpolation (TASI) for voice traffic[4,5] and Adaptive Data Multiplexing (ADM) for interactive data traffic.[6] This concept is illustrated in Figure 3.2-3.

Time Assignment Speech Interpolation (TASI), as referred to in this section, is the process that inserts voice bandwidth into the silent (inactive) bandwidth inherent within speech. In doing so, more efficient use of the transmission media is possible. Similarly, Adaptive Data Multiplexing (ADM) inserts active data bandwidth into the silent bandwidth inherent within interactive data messages. For circuit switching concepts, TASI and ADM activity can operate concurrently because TASI takes place only on silent voice channels and ADM only on inactive data channels. Both TASI and ADM can be applied to the ILRAN/TSN network either on a link-by-link or an originating-node-to-terminating-node (i.e., tandem TASI/ADM) basis as depicted in Figures 3.2-4 a and b. The link-by-link TASI/ADM models were used in the analysis.

3.2.2.2 Packet Switching

In packet switching, messages are divided into segments of certain maximum size, called packets, for transmission through the network. In addition to message information bits, a packet contains information bits to route and control it through the network.

Packets can be thought of as envelopes into which data is placed; the packet header contains addressing and other control information. Transmission through the network is based on the content of the packet header, and thus does not interfere with the data inside the envelopes. The system could be designed with security safeguards to prevent the network computers from prying into the contents of the envelopes.

A packet format normally includes flags, packet header, message content, and error control code (such as Cyclic Redundancy Code) fields. The flags are used to indicate the beginning and the end of control. The error code field is used for detecting errors that may have been introduced during transmission.

Packet format for voice and data can be identical. The requirement for error control of voice packets, however, appears to be less stringent for two reasons. First, because of the redundancy inherent in voice, it can generally tolerate a higher error rate than can data.

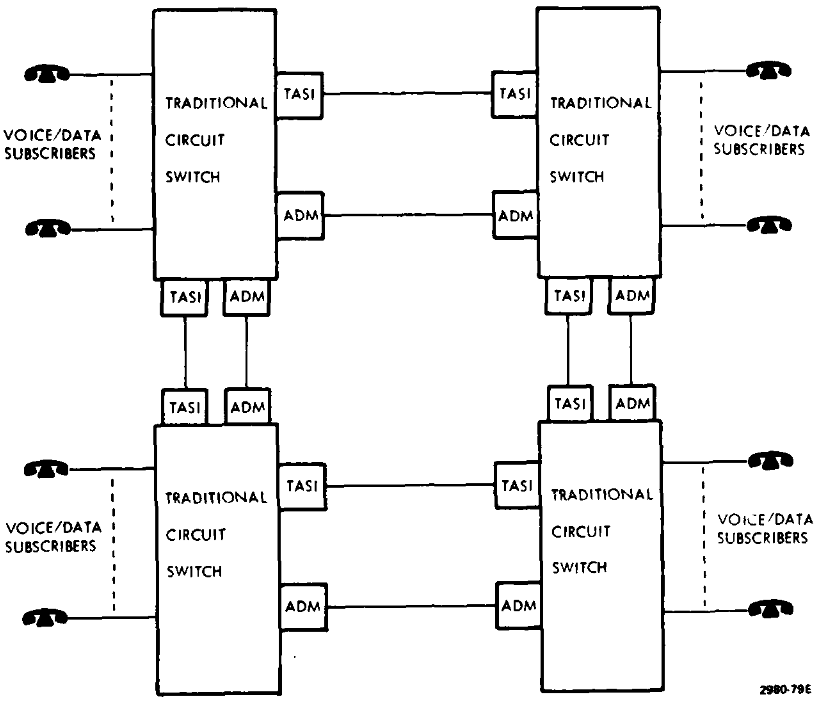


Figure 3.2-3. Enhanced Circuit-Switching Concept

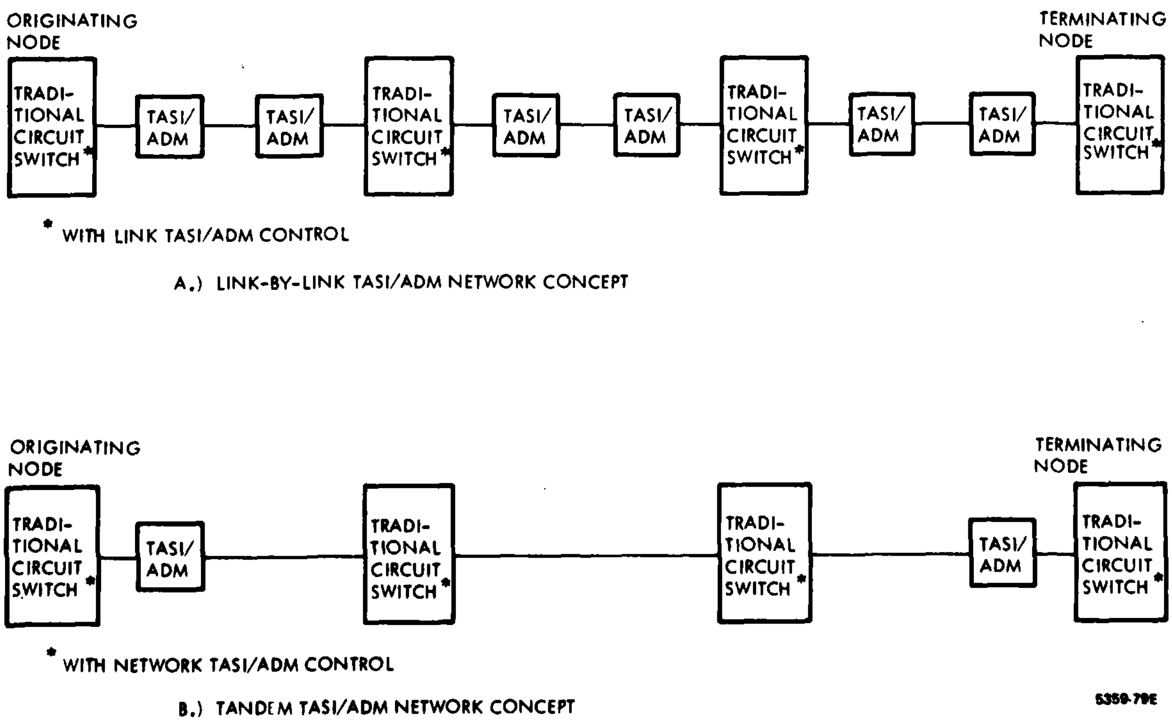


Figure 3.2-4. TASI/ADM Circuit-Switched Network Concepts

Second, when an error is detected, and retransmission is activated to recover from the error as it is done for data, the delay for voice may increase beyond a tolerable level. Thus, error control codes are not needed in a normal communication link with a low bit error rate.

The distinguishing characteristic of the packet switching concept is that the critical network resources, such as bandwidth and buffer space, are not preallocated to any specific user. The resources are dynamically used to satisfy the needs of individual users at any given instant.

There are three basic packet switching concepts to integrate voice and data. These concepts are essentially based on fundamental variations in packet transport protocols. The three concepts are briefly discussed in the remainder of this section.

3.2.2.2.1 <u>Packet Virtual Circuit (PVC) for Voice and Data</u> - Under the PVC approach, a route is established through the network during a call-initiation period, and the same physical path is used by all packets for the call. At the call-initiation time, a signaling packet, (i.e., "call request" packet, is originated by the source switch and sent to the destination switch where, after verifying that a virtual circuit path can be established for the voice or data call, a call confirmation packet is originated and returned to the source switch. Along the path, data structures are setup at each switching node to specify the outgoing link for every call. All subsequuent packets during the call or data transfer transit that fixed path. During the call or data transfer, the buffer capacity is not reserved implicitly (see Figure 3.2-5). Upon termination of the connections, the path is released.

The fixed path strategy guarantees that the packets will arrive at the destination in sequence. It also requires a virtual connection, analogous to that of circuit switching, to be setup between the source and destination, and maintained for the duration of the conversation. It can reduce the amount of processing required for individual packets at intermediate switches since repetitive information, such as call identification and routing, could be stored at intermediate nodes. A simplified header, indicating the call to which the

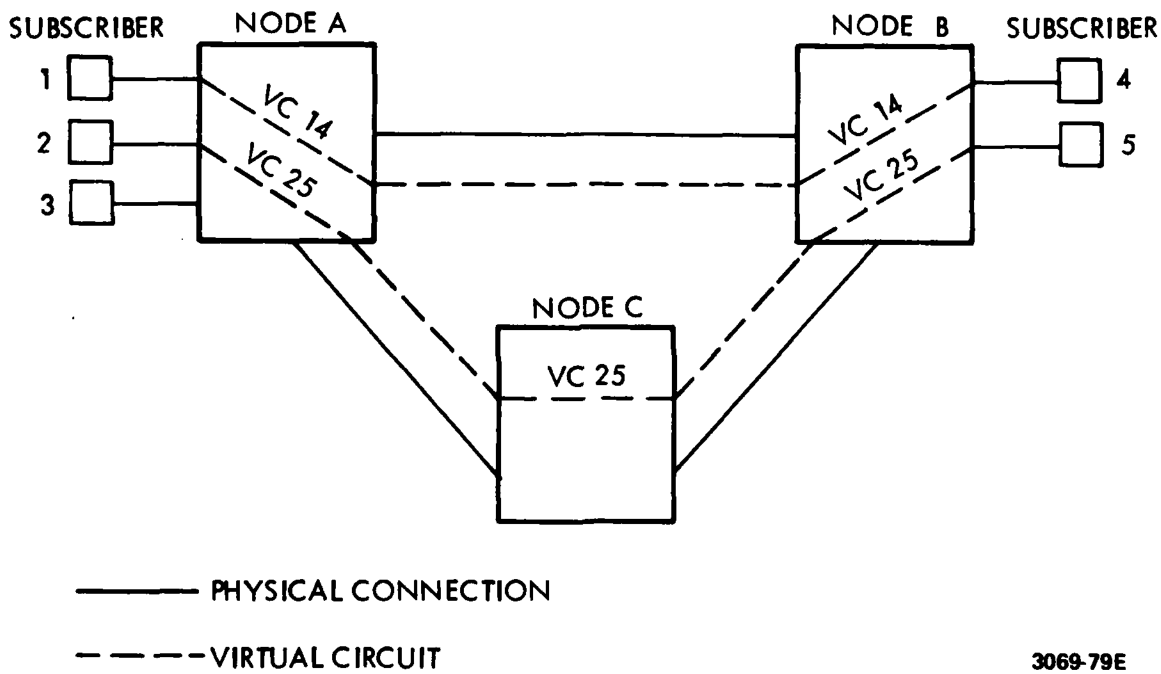


Figure 3.2-5. Virtual Circuit Example

packet belongs, could be sent, which is sufficient for the receiving node to access the prestored information needed to process that packet. The simplified header can further improve delay performance and shorten the time required to create packets. An added overhead in the fixed-path approach is that the virtual circuit needs to be reestablished every time packets are rerouted, (e.g., in the presence of link and node failure).

3.2.2.2.2 <u>Packet Adpative Routing (PAR) for Voice and Data</u> - In the PAR concept, voice and data packets are independently routed to the destination. No path is setup for the duration of the call. Each packet is transported across the network to its destination independently of other packets for the same destination. Individual packets can be alternately routed as appropriate. This approach is a "pure" packet network, and could result in packets belonging to the same talk spurt being received out of order because of the varying delays encountered over different network paths. Hence, a resequencing mechanism is required at the destination node or in the access area.

The PAR concept requires larger packet headers than those in the PVC concept, since each packet carries all necessary information to independently route it through the network. Such increase in packet header is expected to result in reduction in the maximum achievable information bit throughput of the network when compared with similar results for the PVC network under identical conditions. One advantage of the PAR concept is its responsiveness to failures in the network by routing packets around the failed paths.

3.2.2.2.3 <u>Virtual Circuit for Voice, Adaptive Routing for Data (MIXD)</u> In the MIXD approach, voice packets are routed over virtual paths as in the PVC scheme, and data packets are independently routed as in the PAR concept. This is a combination of the Virtual Circuit and Adaptive Routing approaches, and attempts to associate each routing strategy with the class of data for which it is best suited; hence, voice packets use the fixed-path protocol, where data packets use the path-independent protocol. A disadvantage of the approach is that two separate protocols are required.

A detailed description of the MIXD concept follows.

The MIXD concept transmits voice traffic and data traffic in packetized form, but treats the voice packets and data packets with different protocols that are tailored to the special characteristics of each kind of traffic. When a voice subscriber requests a connection, a fixed-path protocol establishes a virtual circuit as described previously. The protocol considers the available bandwidth on each trunk when making the routing decisions. (In order to guarantee some level of service to the data traffic, the protocol may be allotted some fraction of the available bandwidth on each trunk.) The requesting subscriber specifies an initial bandwidth, which is used in determining capacity requirement. If no path can be found in which all links have sufficient available capacity, then the connection is blocked and the subscriber is notified. The connection may have the capability to preempt other connections and thereby make bandwidth available for its use. A military network normally has a priority scheme to give better service to certain data subscribers (lower delay).

Once the virtual circuit has been established, the traffic transfer functions as in the PVC scheme described previously. The individual voice packets carry minimal header information and are routed along fixed routes to the destination. The voice packets would be given priority on the trunks over the data packets within the constraints of the traffic precedence scheme. The virtual circuit is disestablished as described previously.

When data traffic is presented for transmission, the data packets are furnished with complete headers, and follow a path determined by adaptively updated routing tables in the nodes. In each node, the data packet is assigned by the routing table to a trunk queue that is managed by precedence schemes. The trunk choice is based on both the destination address and the network congestion and connectivity. As these change, different packets from the same source may follow different paths to the same destination.

3.2.2.2.4 Packetized Speech Model - The packet switching technique for handling voice traffic has inherent characteristics to support a TASI-like advantage. In the circuit switching technique, the TASI advantage is defined in terms of the ratio of the number of callers supported to the number of circuits required to maintain the cut-out fraction below a prescribed threshold. The packet switching technique provides another mechanism that offers a TASI-like advantage. Since packet switching involves the buffering of information, the speech information from a caller can be stored in a buffer when the backbone circuits are temporarily busy (i.e., in use), and then transmitted over the backbone when the circuits are available. Unlike the cut-out approach in circuit switching, delay management in packet switching can provide a TASI-like advantage. Variable delay is an inherent property of a packet switching network; however, the variation in delay above a certain minimum has a pronounced impact on the quality of speech at the destination. Thus, like circuit switching, a TASI-like advantage can be obtained at the expense of speech quality. Variation in the delay can be minimized by appropriate design decisions for protocols, flow control, precedence scheme, and speech reconstitution algorithm at the destination.

Voice and data bit streams are segmented into the maximum-size packets, transported through the network as packets, and delivered at the destination in bit streams. Depending on the architecture, the conversion of bit streams to packets and vice-versa may occur in the access area or in the switching nodes. Three basic switching concepts (PVC, PAR, and MIXD), as described in Paragraph 3.2.2.2, are employed in transporting packets through the network. In each concept, the speech activity detector does not send a segment to the switching node whenever it detects silence. Thus, network resources are not used during the silence periods. At the destination, the nodes deliver segments containing speech information. A speech reconstitution algorithm at the destination can use a technique to minimize variation in delay, and the voice information can then be delivered to the destination.

3.2.2.3 Hybrid Switching

In the hybrid switching concept, switching and transmission facilities of the network are shared between traffic, with a single facility providing both circuit and packet switched modes of operation. The transmission capacity of the link between two nodes is used to carry circuit switched traffic at one instant, and packet switched traffic at another instant. There are several variations of this approach. In one variation, voice and other real-time traffic is circuit switched while interactive data and other non-real time traffic is packet switched. Another alternative is to packet switch the voice traffic, and circuit switch bulk data and facsimile, with any necessary error control for the circuit switched data provided via packets or a small transmission capacity in the reverse direction. At the switch, the programs and functions needed to perform either circuit or packet switching are present. The switch processing capacity is dynamically shared by these functions; the load at any instant is dependent on the traffic mix, routing, etc., at that time.

Figure 3.2-6 illustrates the approach for the case where the voice is circuit switched. Wideband digital trunk capacity is divided into continuous, constant-period, master frames in the time domain. The period (e.g., 10 or 20 ms), once selected, would be constant through the network. Within a frame, portions are utilized as (1) a marker to indicate the start of a frame, (2) a region that multiplexes those synchronous types of traffic normally associated with circuit switched traffic, and (3) a region that incorporates packet switched or message switched traffic, and is handled as packet switched; interactive data and signaling messages for circuit setup and disconnection would be packet switched; and bulk data would be either circuit or packet switched, depending on the application. Within the frame, traffic of varying bandwidth can be accommodated simultaneously by allocating time shares as needed, until the total capacity of the wideband trunk has been allocated.

In Figure 3.2-6, the period of the master frame is 10 ms and the transmission rate is 1.544 Mb/s. Thus, 15,440 bits are provided within each master frame, with frames transmitted at a rate of 100 per

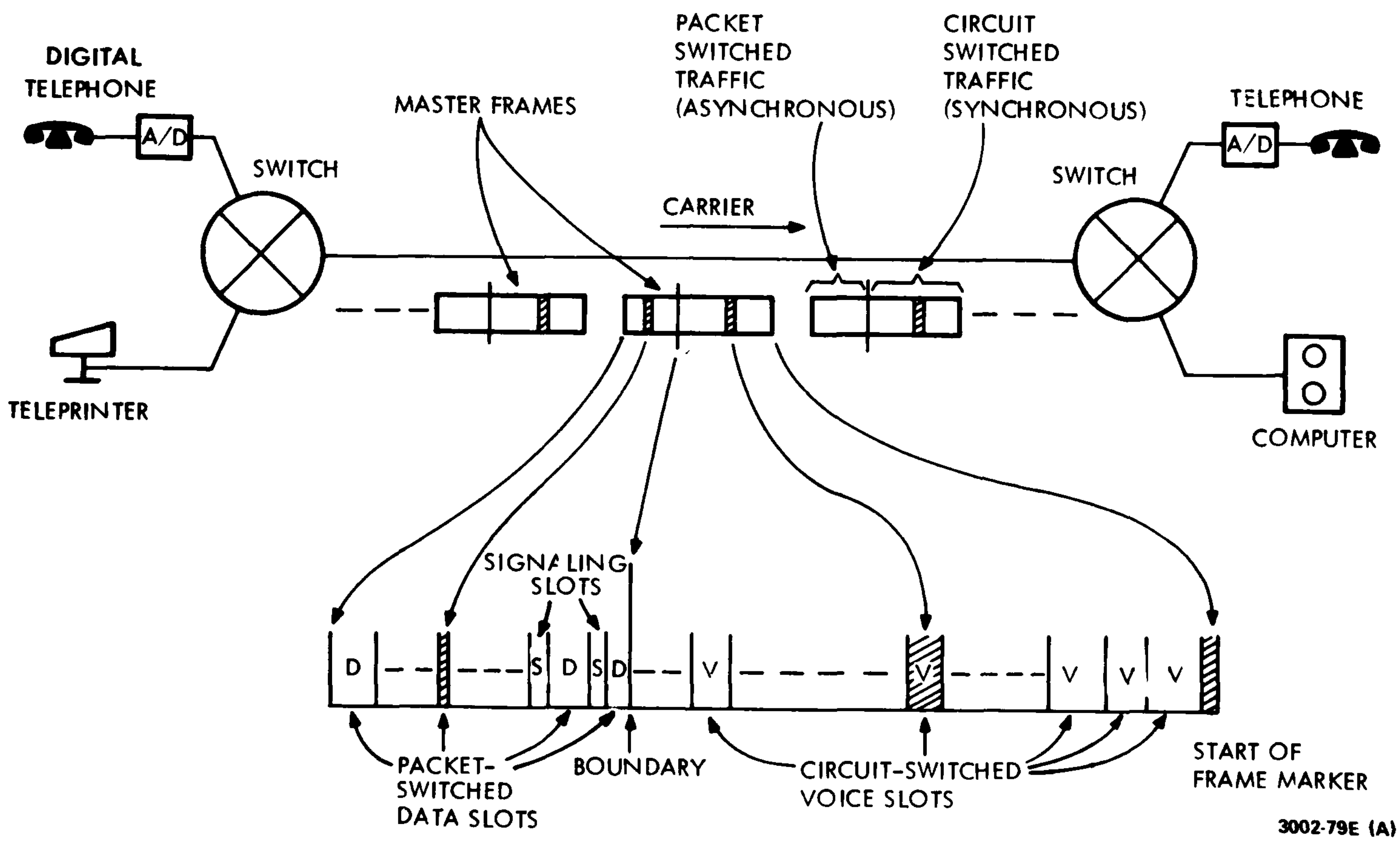


Figure 3.2-6. Hybrid-Switching Channel Capacity Structure

second. Digitized voice at 16 Kb/s would require an allotment of 160 bits per frame for the duration of the conversation. For example, if a call lasts 5 minutes, then 160 bits must be reserved for each of 30,000 master frames (100 frames per second times 300 seconds). Interactive data between a terminal and computer are transmitted in the form of packets. Incorporated into the packet format will be information necessary for routing, security, identity, and precedence.

There are various approaches to the sharing of transmission capacity in a hybrid system. The frame boundary is software controlled and the partitioning of frame capacity to circuit switched or packet switched regions need not be fixed or identical on different network links. In the following sections, variations of this basic generic approach will be investigated. The impact of the basic hybrid technique on transmission efficiency and delay will be assessed, and the effect of variations used to further increase channel utilization will be evaluated. As we shall see, the various hybrid approaches, and in general the variations of integrated voice/data concepts within the circuit and packet switching concepts, are not necessarily discrete and exclusive, but rather are entries on a continuum of techniques in which performance and control blend from one technique to another.

3.2.2.3.1 SENET (Slotted Envelope Network) Approach - In the SENET approach[7], voice and other real-time communications (e.g., facsimile) are transmitted by circuit switching techniques in the synchronous traffic (or Class I) portion of the frame and interactive data by packet switching techniques in the asynchronous traffic (or Class II) portion of the frame. Bulk data may be transmitted either as Class I or Class II traffic. A movable boundary arrangement allows both types of traffic to coexist on the same link according to a role by which one type of traffic can utilize idle channel capacity normally assigned to the other.

Figure 3.2-7 shows (for illustrative purposes) the channel map allocations for a typical SENET frame. The Class I region of each frame is specified by a start-of-frame (SOF) marker and by allocation maps maintained at both ends of each link. These allocation maps are

TYPICAL MASTER FRAME CLASS II ALLOCATIONS

ALLOC. START	ALLOC. WIDTH	CALL IDENTITY AND PRECEDENCE	ALLOCATION PURPOSE [TYPICAL]
9896	8	CLASS I/II BOUND	8-BIT ADCCP FLAG SEQUENCE
9904	112	CCIS-1:Y	CALL ANSWERBACK, RELEASE, PREEMPT RELEASE, OPERATOR RECALL
10016	104	CCIS-2:Z	ACKNOWLEDGE, GLARE, OUT-OF-SERVICE SYNC, CALLED PARTY UNAVAILABLE, INVALID NUMBER, INCOMPATIBLE CONN.
10120	752	PACKET-1:Z	TYPICAL TRANSMITTED INTERACTIVE, QUERY, DATA BASE UPDATE - SINGLE PACKET MESSAGES
10872	712	PACKET-2:O	TYPICAL TRANSMITTED/RECEIVED: NARRATIVE/RECORD LINE BLOCKS [AVERAGE: 11.6 PER MESSAGE]
11584	104	CCIS-3:O	ALL TRUNKS BUSY, EQUIPMENT BUSY, INVALID ROUTE
11688	1080	PACKET-3:P	TYPICAL RECEIVED: INTERACTIVE OR RESPONSE - ONE OF MULTIPLE PACKETS PER TYPICAL INTERCHANGE
12768	224	CCIS-4:P	CALL INITIATE
12992	160	CCIS-5:R	CALL COMPLETE, ACKNOWLEDGE CALL INITIATE
13152	2080	PACKET-4:R	BULK-1 DATA, NON-SENSOR BULK-2 DATA, ARQ-FACSIMILE
15232	208	NONE	IDLE CAPACITY

ADDITIONAL CLASS I ALLOCATIONS

CHAN. ALLOC.	START'G POINT	CHAN. WIDTH	FEC-FAX	FAX	DIGITIZED VOICE: PCM	DPCM	CVSD	APC	LPC	VOCODER
6	7508	640	X		X					
7	8148	500	X							
8	8648	480	X		X	X				
9	9128	320	X			X	X			
10	9448	160	X				X	X		
11	9608	96		X				X		
12	9704	80						X	X	
13	9784	48		X						X
14	9832	40							X	
15	9872	24		X						X

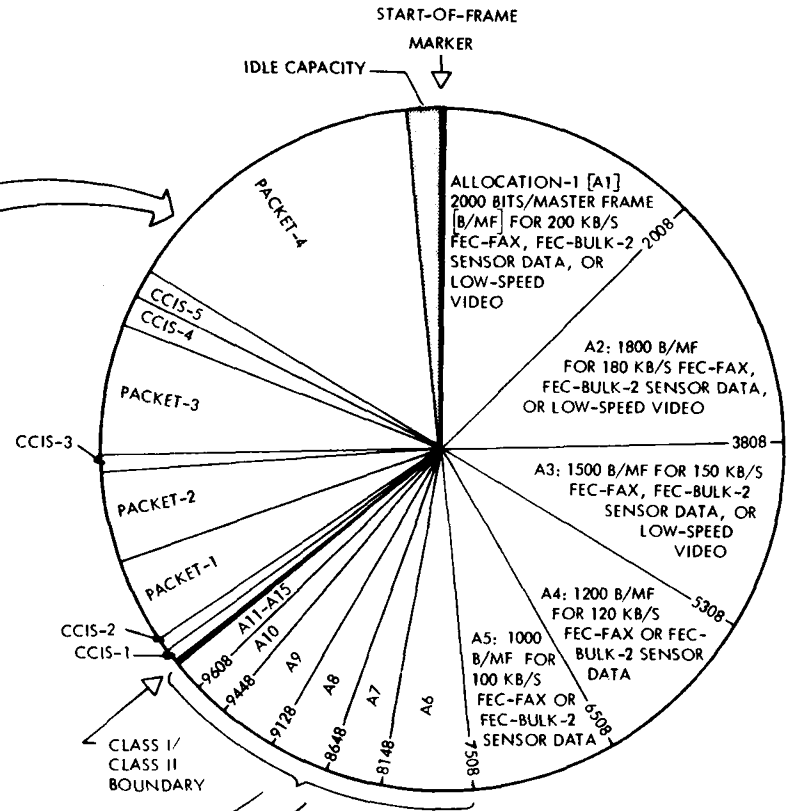


Figure 3.2-7. Typical Master-Frame Map Allocations for the SENET Concept

specific as to call identity, starting points of the allocation in the master frame, magnitude of allocation per virtual connection, and precedence of the virtually connected call. Master frame allocation sequences are maintained in the form of lists; one list per outgoing trunk and one list per received trunk. Connection of a received virtual channel to a transmit virtual channel is by means of list processing.

Virtual channel allocations of each of 15 different rates of transmission of 10 representative kinds of real- and near-real-time communications are shown. Allocations to the Class I region are always made following the last previous allocation (i.e., at the Class II end of the Class I region), thus reducing CCIS signaling to that concerning the allocated call. As calls terminate, all subsequent allocations are incremented toward the SOF marker by the magnitude of the terminated allocation, based on the terminating CCIS message; that is, a CCIS Message directly canceling a single call will result in reallocation of all calls occurring further away from the SOF marker than did the terminated call, without the necessity for individual CCIS messages for each of these reallocated calls. As a result of these call terminations and dynamic reallocations, the position of the Class I/II boundary will vary dynamically, but will always vary in such a direction as to require the minimum size for the Class I calls currently being serviced, and will thereby provide the maximum amount of master frame capacity for other than Class I transmissions.

The Class II region of a master frame is essentially self-identifying in that its contents are individually recognizable packets enveloped in accordance with ADCCP procedure. Consequently, at the first occurrence of an ADCCP flag sequence following the SOF marker, it may be assumed that the Class II region has been entered. This may be cross-checked by integration of the total allocation for Class I with respect to the individual width per allocation. Allocations in the Class II region are made by FIFO with respect to precedence as long as Class II space is available.

There is no boundary marker to signify end of frame. The SOF of each successive master frame serves as the end of the frame of the

preceding master frame. Should the contents of a master frame not occupy all of the available capacity in that frame, the remaining space following the end of the last complete packet in Class II will be filled with synchronous characters (e.g., ADCCP flag sequences).

3.2.2.3.2 Flexible-Hybrid Approach - One of the disadvantages of the SENET approach is that transmission efficiency suffers as a direct consequence of the inefficiency in transmitting circuit switched voice. TASI, which will be examined in the next paragraph, is one method of improving the efficiency but at the price of adding a considerable amount of complexity. In this paragraph, we will describe a variation of the basic SENET approach that would improve the efficiency performance of the hybrid system considerably.

The approach, which we have termed the flexible-hybrid concept, has bulk data in a circuit switched mode while voice and interactive data are handled in a packet switched mode. Several investigators have suggested this scheme as a potentially cost-effective strategy for integrating voice and data in a common system.[8,9]

The flexible-hybrid scheme is illustrated in Figure 3.2-8. Voice is transmitted as packet switched data without error control via the PVC routing algorithm. Interactive and narrative/record data that require error control are transmitted as error-protected packet switch data via adaptive routing. Bulk facsimile, and burst data information are transmitted in simplex form as circuit switched data, with any necessary error control or acknowledgments provided via packets or a small transmission capacity allocated in the reverse direction. There are several advantages to this method. The bulk data class can be transmitted more efficiently and with less delay using circuit switching since, in general, long messages favor circuit switched operations and short messages favor packet switched operations.[10] This requires modification of the SENET approach to allow bulk data to be transmitted over a two-wire path in simplex form, similar to the way bulk data are transmitted via PVC routing in a packet switching scheme. The price to be paid is asymmetrical link maps and the necessity to provide error control in the reverse direction. The SENET scheme specifies symmetrical link maps for allocation of data

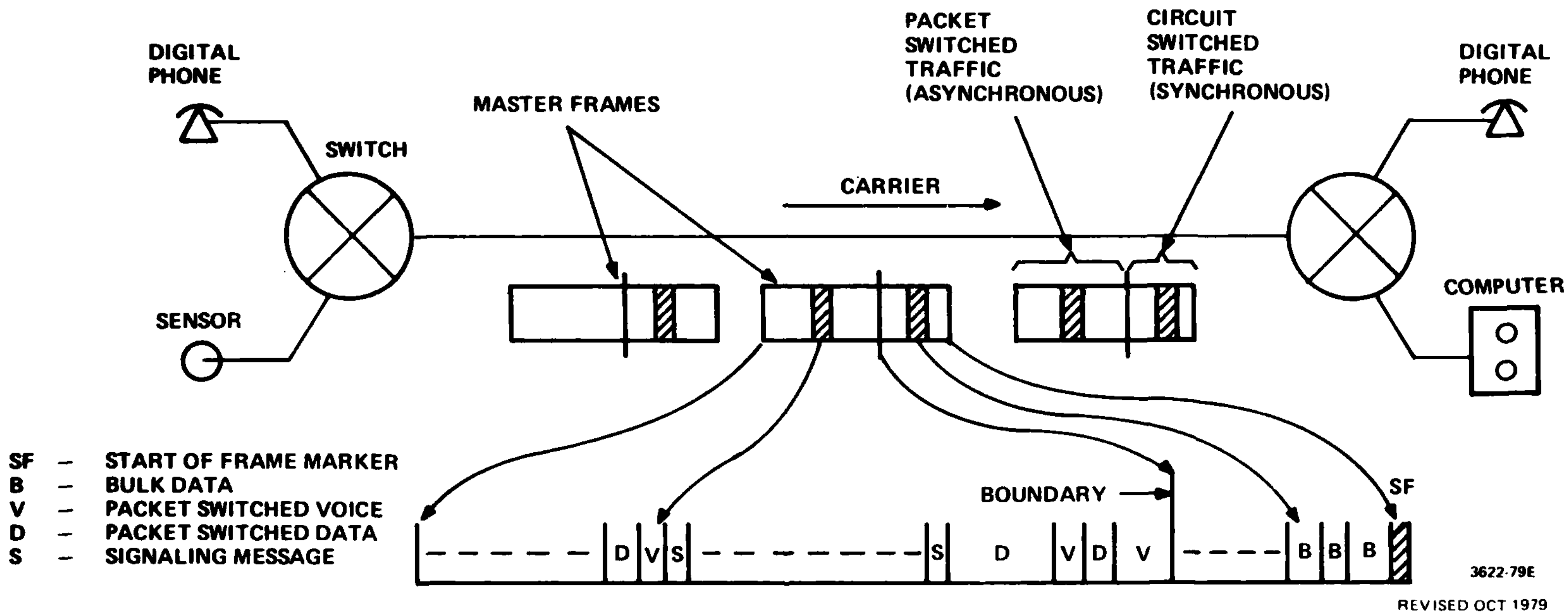


Figure 3.2-8. Flexible-Hybrid Channel Capacity Structure

regardless of direction of transmission (e.g., see Figure 3.2-7). The simplex transmission of bulk-like data would necessitate separate frame map pairs at the ends of each link - one pair of maps for each direction. As stated previously, error control could be provided either through circuit switched allocation of a small control channel in the reverse direction, or via packets sent in the Class II region.

The packet switching of voice provides a TASI-like efficiency advantage identical to that found in the packet switching systems of Paragraph 3.2.2.2. During speech silences, transmission ceases; hence, the capacity is available for servicing other users. Packet switched traffic or message switched traffic requiring error protection is handled conveniently within the packet switched mode of operation.

There are several modifications to the approach that could be used to improve operational performance. The circuit switched region can be used to carry synchronous encrypted voice or data. In addition, high-priority voice or data that require low delay (e.g., voice over satellite links), or delay that is independent of processor load, or voice traffic requiring significant interaction between subscribers to ensure that the information transmitted is correct, could also be circuit switched.

3.2.2.3.3 Enhanced Hybrid Approach - The enhanced hybrid approach is similar to the basic SENET approach described in 3.2.2.3.1; however, both TASI and Time Assigned Data Interpolation (TADI) techniques are added to increase the efficiency of transmission links.

TASI, as referred to in this section, is the process that inserts voice bandwidth into the silent bandwidth inherent within the SENET Class I voice channels. In doing so, more efficient utilization of the transmission media is possible; however, note that no attempt is made to employ TASI for groups of subscribers of differing voice digitization rates, (i.e., differing Class I slot allocations). Similarly, TADI inserts active data bandwidth inherent within the Class I voice slot allocations. In order not to introduce undue

complexity, it is assumed here that TASI and TADI do not take place concurrently, and that they are applied to the ILRAN/TSN network on a link-by-link basis.

Figure 3.2-9 illustrates one approach to implementing TASI/TADI in a hybrid system. A control vector in the circuit switched portion of every frame indicates which user slots are in talk spurt and which are in silence. These are identified by x's in the diagram. Virtual circuits in which the voice source is in an idle state can be used to carry data or additional speech from a second voice source. The former is designated Time-Assigned Data Interpolation (TADI) and the latter is Time-Assigned Speech Interpolation (TASI). Voice sources that leave the silence state and resume speech generation may be "frozen out", (i.e., they may not find an available slot in which to transmit). A sufficient number of slots must be provided to ensure that the percentage of speech "frozen out" is within an acceptable percentage (e.g., 0.5 percent).

3.2.2.3.4 <u>Variable-length frame/packet (VLFP)</u> - Miyahara and Hasegawa have proposed an integrated multiplex scheme for the hybrid system which will help to eliminate unused frame capacity.[11,12] In an integrated circuit and packet switched multiplex structure, such as the SENET, the communication link is dedicated to circuit switching at one instant and packet switching at another. The frame structure is divided into a circuit and packet switched region. Frame intervals are kept constant for the synchronous assignment of circuit switched time slots. Packets are fit into that part of the frame following the circuit switched region. It is desirable under this scheme to transmit as many packets as possible, and thus minimize the idle capacity at the end of the frame. But, because of the constant frame format, either packets are too long for transmission in that frame, in which case capacity is wasted, or the packet length is shortened, which increases the overhead of that packet relative to its information field; however, if the frame interval is allowed to have variable length, a packet that exceeds the residual capacity of a frame can be transmitted over two consecutive frame intervals. This

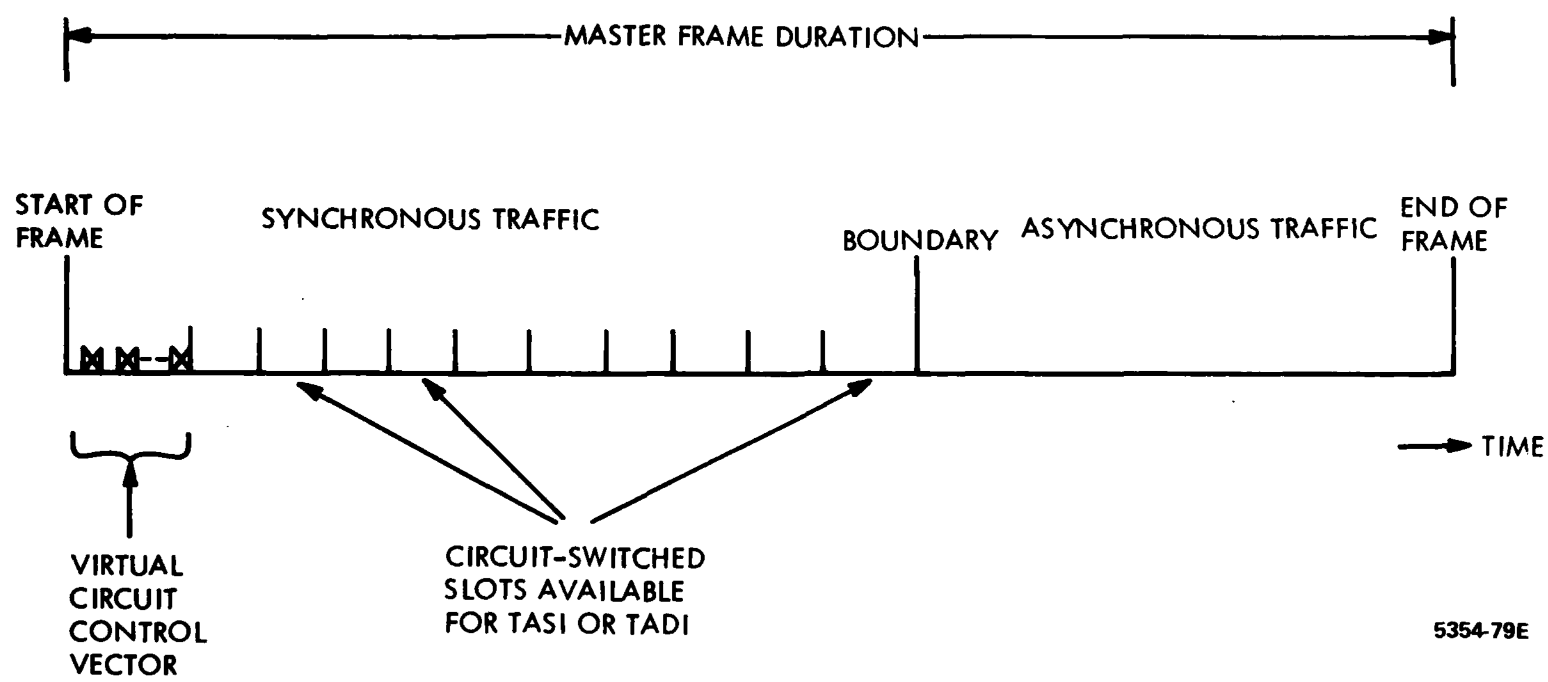


Figure 3.2-9. Channel Capacity Structure Using TASI/TADI Techniques

will result in an increased transmission efficiency, an increased flexibility in handling varying traffic demands, and an improvement in average packet transmission delay.

One problem with this method is that if the packet is being transmitted over two consecutive frame intervals, the circuit switched region cannot interrupt at that time to begin its scheduled transmission. Under these conditions, the transmission delay of the circuit switched portion is no longer constant. The interarrival time of successive circuit switched bit groups will no longer be equal to the constant frame interval (10 ms); thus, the transparency to delay dispersion, which is a natural consequence of circuit switching, would not be preserved; however, if the dispersion of frame interarrival times can be upper-bounded to a maximum value, and that value is sufficient for regaining synchronization at the destination node, and in addition, there is little or no impact on voice quality and intelligibility, then the transparency of circuit switching can still be offered.

Figure 3.2-10 illustrates this process. Frames are scheduled for departure at fixed time intervals; however, the new frame will not be transmitted if a packet is in transmission at that time. Note that, if the traffic on the link decreases sufficiently, the delay dispersion can readjust itself to the nominal value. This is the case for frames 4 and 5 where the actual and scheduled departure times for frame 5 coincide.

3.2.2.3.5 Noncontiguous Frame Format (NCFF) - In this section, we consider a technique introduced by Zafiropulo in which the capacity assigned to circuit or packet switched data in a frame does not necessarily have to be contiguous.[13] In Figure 3.2-10, we depict a single frame in which capacity is assigned to circuit switched data, packet switched data, or idle capacity. Linked lists are used to keep track of which bits are assigned to each type of traffic. Each list has an identical structure and contains the channel address, the location and number of contiguous bits (slots) assigned in the frame, and a link to elements of other lists.

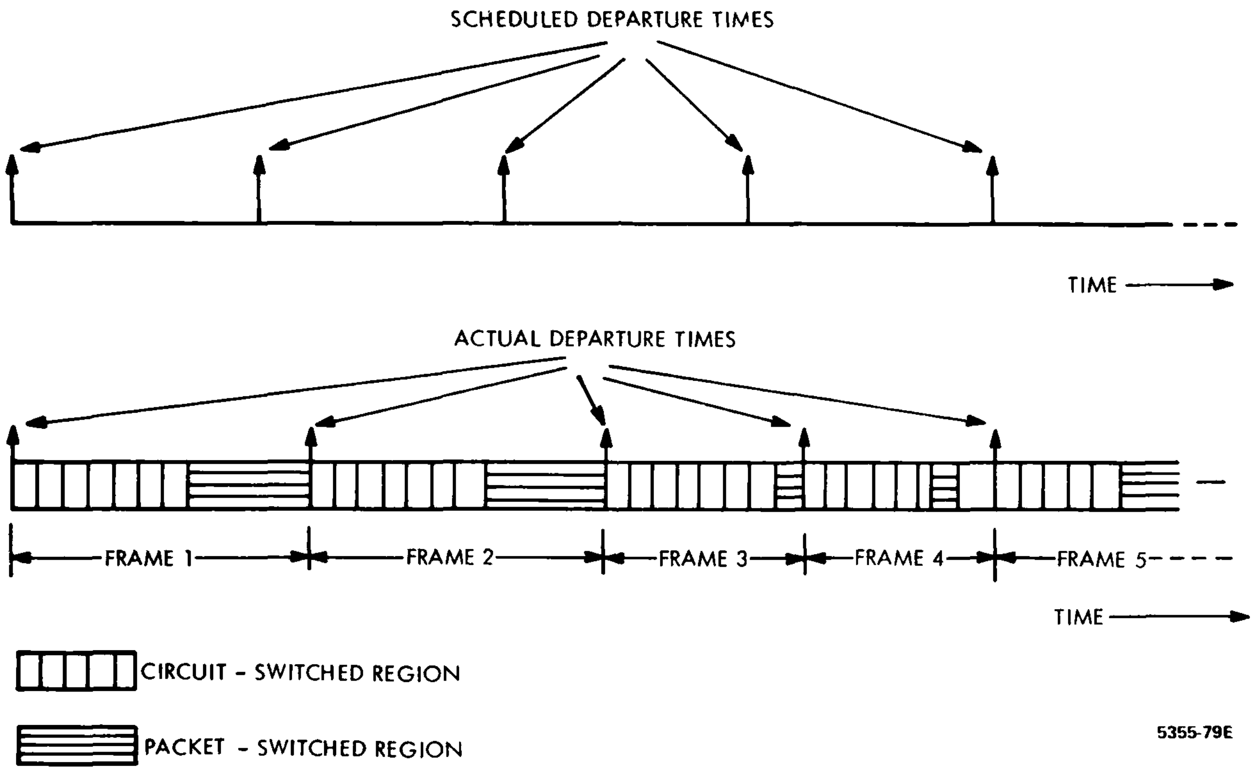


Figure 3.2-10. Variable-Length Frame/Packet Departure Times at a Node

The circuit switched and packet switched lists describe all the slots used by the respective circuit and packet switched channels. The channel address characters of the packet switched list elements are ignored. In this scheme, all slots not allocated as circuit switched slots are potential candidates for carrying packet switched traffic. When a circuit switched channel gets disconnected, the elements of that list are transferred first to a liberate list and then into the packet switched list, if required. Capacity that is not used is assigned to a free list. The circuit switched list is two-dimensional because a majority of noncontiguous bit positions can be used by a single circuit switched channel (see cross-hatched area in Figure 3.2-11. Elements belonging to a secondary circuit switched list always contain the same channel address.

Allocation of a circuit switched channel is as follows: If the width of the circuit switched slot is smaller or equal to the largest packet switched slot, then the packet switched slot yielding the best fit is allocated. If the width is larger than the largest packet switched slot size, then the largest slot is allocated to the channel (see Figure 3.2-12). The remainder is generated, and the best fit procedure is repeated. Allocations are held for the duration of the call, after which they are released for packet switched traffic. Thus, bits may be assigned anywhere in the frame and are not necessarily contiguous.

An advantage of this approach is that a large combination of different circuit switched channel speeds can be accomodated. In addition, the linked list representations can support different channel allocation algorithms without the need to change hardware; however, this approach is much more complicated than the basic SENET approach. The requirement to break up an information block to fit into the available slots requires significantly more decision logic than a technique that allows each information block to be transmitted as contiguous data. An interesting property of the noncontiguous approach is that, during call setup and termination, only the bits associated with a particular call are affected. The remainder of the frame remains intact. In the SENET scheme, when a circuit switched

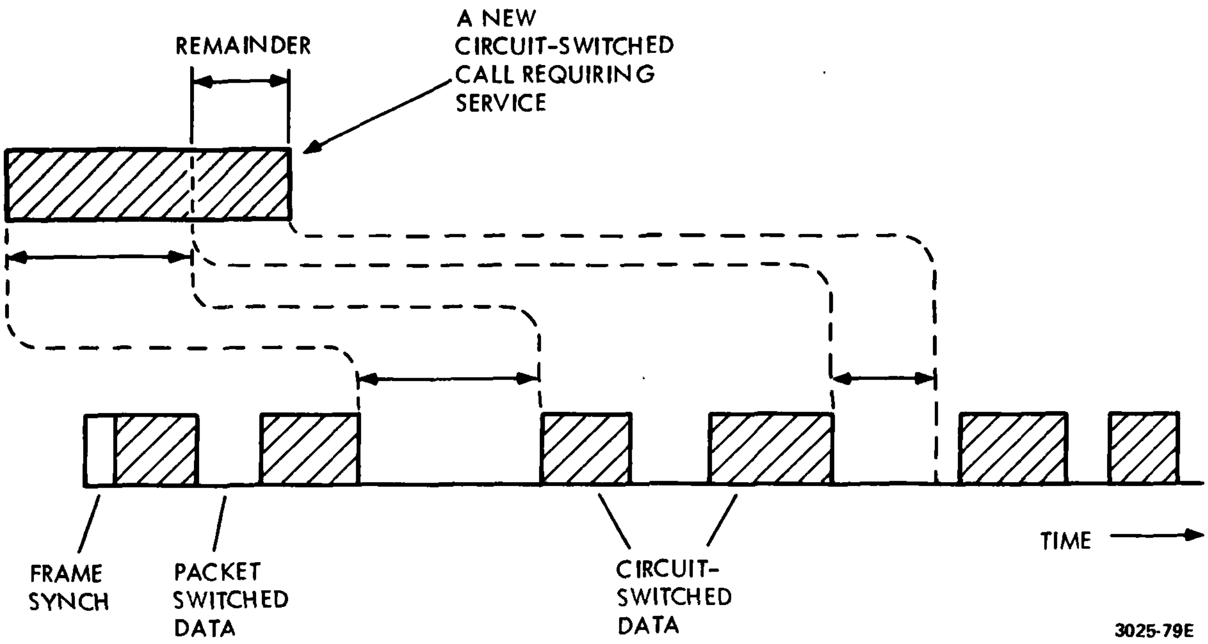


Figure 3.2-11. Flexible Multiplexing of Circuit - and Packet-Switched Data Traffic

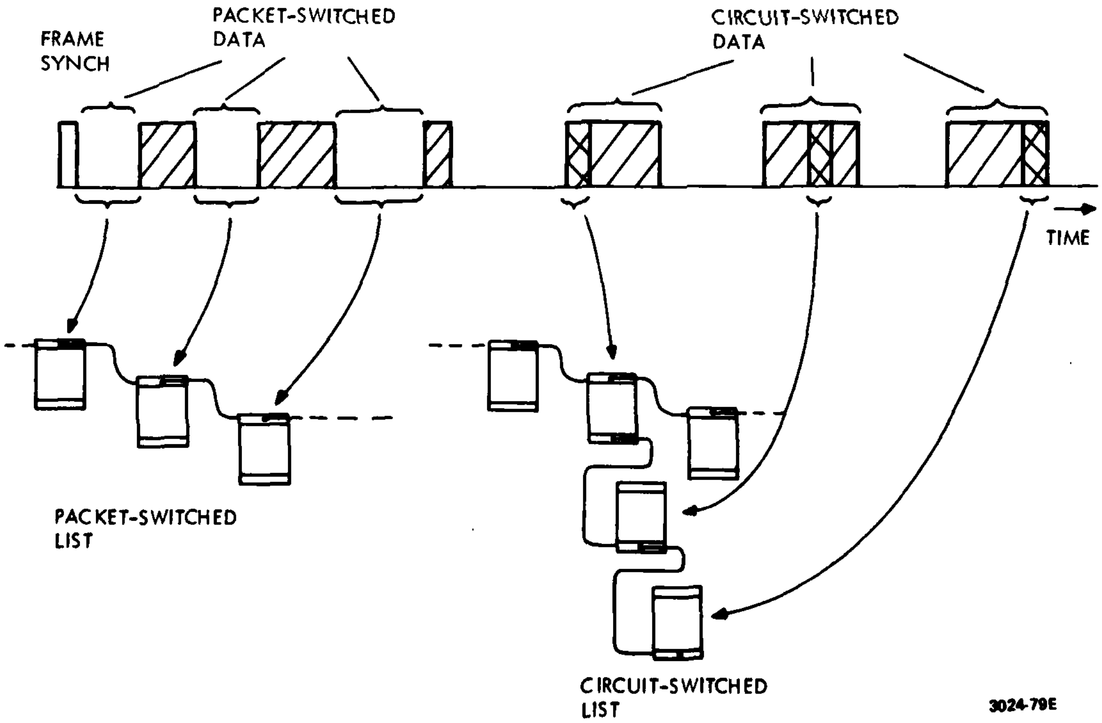


Figure 3.2-12. Algorithm for Allocating a Large Circuit-Switched Slot

call terminates, the frame is recompacted, i.e., those calls in the frame following the terminated call are shifted up to close the gap. Because of this, meticulous control of the maps in the SENET scheme is necessary. Transmitting and receiving switches must agree exactly on what is located where, how much of the frame each allocation requires, and the exact timing of changes in the allocation maps. The resulting coordination complexity hampers the application of a TASI technique to interpolate speech into the silent periods of other speech. This complexity, however, does not exist for the noncontiguous approach since the frame is not recompacted after each call termination. Therefore, application of a TASI-like technique should be easier using the noncontiguous approach than with the SENET approach.

3.2.3 Evaluation Results

The concept evaluation effort examined the transmission efficiency of each AISC concept to determine its capability for making full use of the system's transmission capacity. Two types of transmission efficiency were studied; overhead efficiency and throughput efficiency. Overhead efficiency provides a measure of the number of information bits that can be transmitted on the line relative to the amount of overhead bits required. This represents a maximum achievable efficiency, i.e., the efficiency obtainable with a solidly packed line. The throughput efficiency takes into account the stochastic arrivals of packets, their lengths, and the effects of loading at the queues. It provides a more conservative estimate of efficiency, but is a value closer to that which would be obtained in a realistic environment. There are other definitions of efficiency that may also be used to evaluate the different concepts.[16] These particular models, however, provided the best analytical approach for comparative evaluation of the variations of circuit, packet, and hybrid switching concepts.

Figures 3.2-13 and 3.2-14 show the overhead and throughput efficiencies for the various AISC concepts over a range of voice/data distribution ratios. Figure 3.2-13 presents the results of overhead

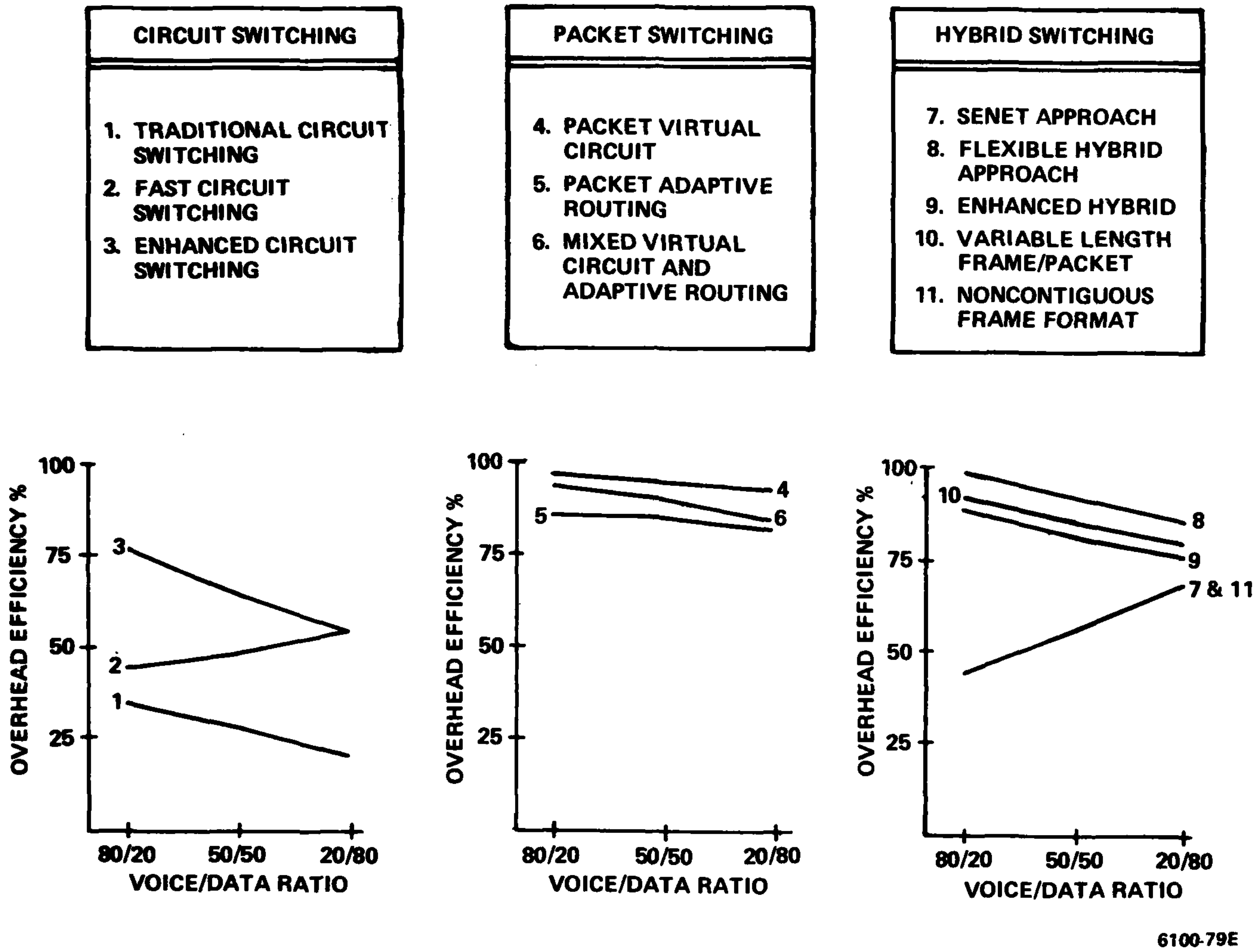


Figure 3.2-13. Comparison of Integrated Voice/Data Concept as a Function of Transmission Overhead Efficiency

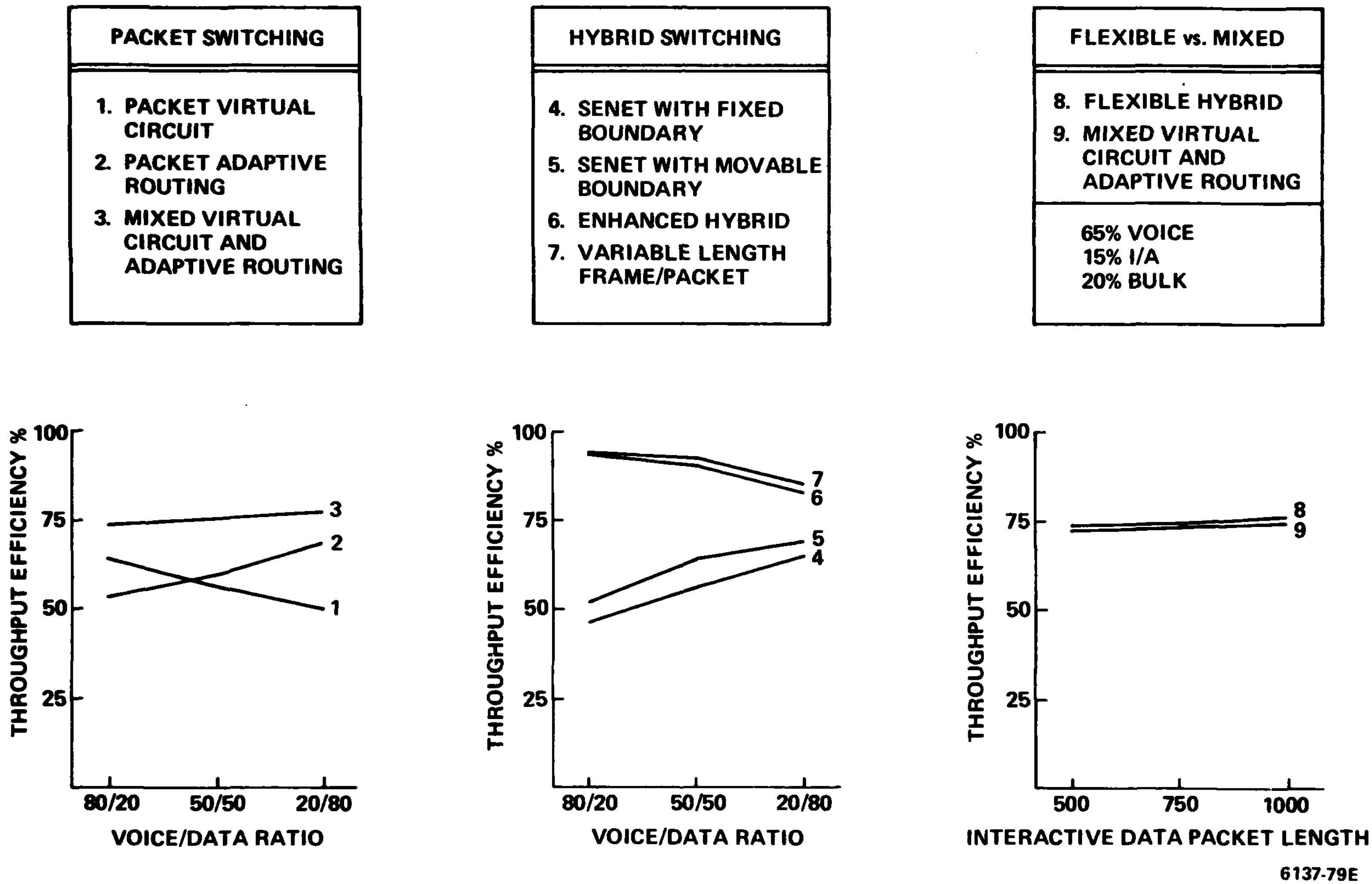


Figure 3.2-14. Comparison of Integrated Voice/Data Concepts as a Function of Throughput Efficiency

efficiency for the three generic switching concepts; circuit, packet and hybrid. Figure 3.2-14 presents the throughput efficiency for the packet and hybrid switching concepts. It also compares the Flexible Hybrid and MIXD packet switching systems for a specific distribution of 65% voice, 20% bulk data, and 15% interactive data, based on prediced DCS traffic for the mid-1980s.

3.2.3.1 Transmission Efficiency

a. Circuit Switching

Traditional circuit switching is not very efficient (20 to 35%), particularly for mixes of voice and data that specify large amounts of interactive traffic. The transmission efficiency for fast circuit switching is better than that for traditional circuit switching (on the order of 50%), because of the elimination of idle time for data calls. A significant improvement is seen for the case of enhanced circuit switching, in which TASI is used for voice traffic, and Adaptive Data Multiplexing (ADM) is used for interactive data traffic. Here, transmission overhead efficiency varies from 55% at a 20/80 voice/data ratio to 75% at an 80/20 voice/data ratio. Also, system complexity is comparable to that of the PVC packet switching approach; however, there are some serious drawbacks to the enhanced circuit switching approach. First, data cannot be managed by delay, but must be treated with a loss strategy. Thus, overall network efficiency decreases, because flow control cannot be stochastic in nature with controlled delay and alternate routing and degenerates into ordinary traffic load control using "throttling" techniques. In addition, the preemption of low-precedence voice and data called by high-precedence data results in an increase in data retransmissions and voice call retries. Another disadvantage to enhanced circuit switching is that link error cannot be applied to data. Only user-to-user error control will be possible, and with more than a minimal amount of noise present retransmission will place a heavy burden on the network.

b. Packet Switching

The overhead efficiency for packet switched voice and data (see Figure 3.2-13) is best for the case where both voice and data traffic are sent by packet virtual circuit (PVC). For this case both the voice and data headers are taken to be 32 bits, reflecting the reduction in header information due to fixed routing of packets and storage of packet transmission information at intermediate nodes rather than in individual packets. Data packets are assumed to be 1000 bits in length. PVC has a further advantage in that it requires the implementation of one transport mechanism for

voice and data rather than two. Delay performance is good because of the smaller packet headers required; however, a disadvantage is that data cannot be adaptively routed to its destination in the event of temporary congestion. An adaptive routing algorithm would seek and establish a path through the least congested link and nodes of the network. This would manifest itself in a significant delay improvement for the packet adaptive routing (PAR) strategy and the virtual circuit for voice/adaptive routing for data (MIXD) strategy. It accounts for the somewhat poorer throughout efficiency of the PVC method relative to the other two concepts. (See Figure 3.2-14). An adaptive routing algorithm could also be used when a breakdown occurs in a link or node. In this case, the PAR or MIXD concepts would be able to establish an alternate path dynamically, which would not be possible with the PVC concept.

For packet adaptive routing of voice and data (PAR) the overhead and throughput efficienty drops considerably. The reason for the decrease is that more overhead is required to route voice and data packets adaptively at each node than is required for a fixed-path scheme. For this analysis, packet headers were taken to be 150 bits, similar to that used in AUTODIN II. The "pure" packet approach, with dynamic routing of voice and data, yields the most significant voice delay. This is because the longer the packet overhead, the more uniform the supervisory proceeding, and the better multirouting can be applied, but the longer the packet delay; however, PAR is also the least complex of the packet switching concepts, since it is most like the packet switching systems in use today.

The MIXD system is an intermediate approach. Voice is sent by virtual circuit while interactive data is routed adaptively. Bulk data can be sent by either method, although it would typically be fixed path routed. The PVC traffic, (e.g., voice has a 32-bit header while the data traffic is sent PAR with a 150 bit header). Overhead efficiency results fall somewhere between the other two concepts; however, throughput efficiency is considerably higher for the MIXD concept. The presence of adaptively routed traffic means that the adaptive routing algorithm will have the capability to utilize the bandwidth of the entire network; hence, actual efficiencies for the MIXD system are closer to the upper limit of overhead efficiency presented here. The overhead efficiency for all PVC traffic is higher than that for the other concepts, but the overhead efficiency is only an upper limit. In general, the PVC concept does not permit a distribution of traffic in the network that gives the most efficient use of the trunks. Connections can be blocked when a few critical links are loaded while the other links are only lightly loaded. There is no simple analytical method to include this last factor in these curves. The PAR concepts theoretically permit optimum use of network resources; however, the overhead

efficiency, the upper bound on the actual efficiency, is much worse for this case. The MIXD concept provides overhead efficiencies between the others with a capability to capitalize on the use of different transport mechanisms to send traffic. The result, however, of having to account for multiple transport mechanisms makes this the most complex of the packet switching schemes.

The same analysis was performed using a voice header of 64 bits instead of 32 bits. As expected, the results were similar except that slightly lower efficiencies were obtained for the PVC and MIXD methods.

c. Hybrid Switching

The efficiency for the hybrid switching models is also shown in Figures 3.2-13 and 3.2-14. In a Flexible Hybrid approach, voice is transmitted as packet switched data without error control via the PVC routing algorithm; interactive and narrative/records are transmitted as error protected packet switched data; bulk and burst data information are transmitted in simplex form as circuit switched data, with any necessary error control or acknowledgements provided via packets or a small transmission capacity allocated in the reverse direction. The Flexible Hybrid scheme achieves overhead efficiencies comparible to, or greater than, any of the packet switching schemes (see Figure 3.2-13) and throughput efficiencies similar to those of the MIXD system (see Figure 3.2-14, Flexible vs MIXD). However, like the packet schemes, it is also susceptible to voice rates with efficiency dropping about 10% in going from 16k bps to 2.4k bps for an 80/20 voice/data mix, and 2 to 3% for a 20/80 voice/data mix. Although among the least complex of the hybrid systems, in terms of the number of functions required for implementation, the Flexible Hybrid is still more complex than any of the packet switching systems, chiefly because of the need to incorporate both circuit and packet switching techniques in the same system.

The SENET hybrid concept shows only a fair transmission overhead efficiency over the full range of voice rates, varying from 44 to 68% over the range of voice/data ratios (see Figure 3.2-13). This is because the synchronous (voice carrying) portion of this hybrid approach is very inefficient, being on the order of 40% However, introduction of TASI/TADI techniques (Enhanced Hybrid) improves the efficiency considerably, especially when the proportion of total bandwidth capacity allocated to voice is high. Improved efficiency is on the order of 80 to 90%. The variable length frame and packet (VLFP) approach further increases the efficiency by another 5%. In addition, these efficiencies are relatively flat over the range of voice/data ratios considered and a broad range of voice rates. Introduction of TASI in the enhanced hybrid and VLFP system achieved throughput efficiency greater than that of

any of the other concepts studied (see Figure 3.2-14). However, tandem TASI techniques imply a large amount of control overhead, far in excess of that needed for the link-by-link transfer of information. The results presented here on efficiency are for the single link case only. Improvements, as a result of using TASI techniques, may be optimistic when applied to a multi link network, because of the additional control required.

Thus, the hybrid system can incorporate advanced techniques, such as TASI and VLFP, which will accommodate different mixes of voice and data traffic with a certain degree of predictability. These techniques can improve the transmission efficiency to levels comparable to, or greater than, those of packet switching; however, data delay behavior in the hybrid schemes is worse than that for packet switching systems. In addition, the complexity of hybrid systems in terms of the number of functions required for implementation is higher than that of packet switching systems; especially where tandem TASI techniques are required.

3.2.3.2 Delay Versus Throughput as a Function of Single Traffic Type

Three single-traffic type conditions were examined in order to determine the region(s) of traffic mix over which a particular concept is best suited. Figures 3.2-15, -16, and -17 show the delay-throughput relationship for the various concepts as a function of single traffic types.

Figure 3.2-15 shows the delay-throughput relationship for a T1 carrier transmitting pure voice. Both MIXD and Flexible Hybrid techniques reduce to the PVC. They show superior delay performance to that of PAR over the whole throughput range. All SENET schemes reduce to TDM. In this case, it is observed that at the SENET start of frame instant, synchronous trunk transmission starts immediately with data extracted from input memory. The total time taken by a voice parcel between the start of frame and the completion of transmission is a random variable that has its values in the range (t, T) where t is the SENET slot duration and T is the SENET frame period. This time takes the lower value if the voice call is assigned to the first slot, and it takes the higher one if the call is assigned to the last slot in the frame. For a voice conversation picked at random, the expected value of the total delay in front of the transmission trunk would,

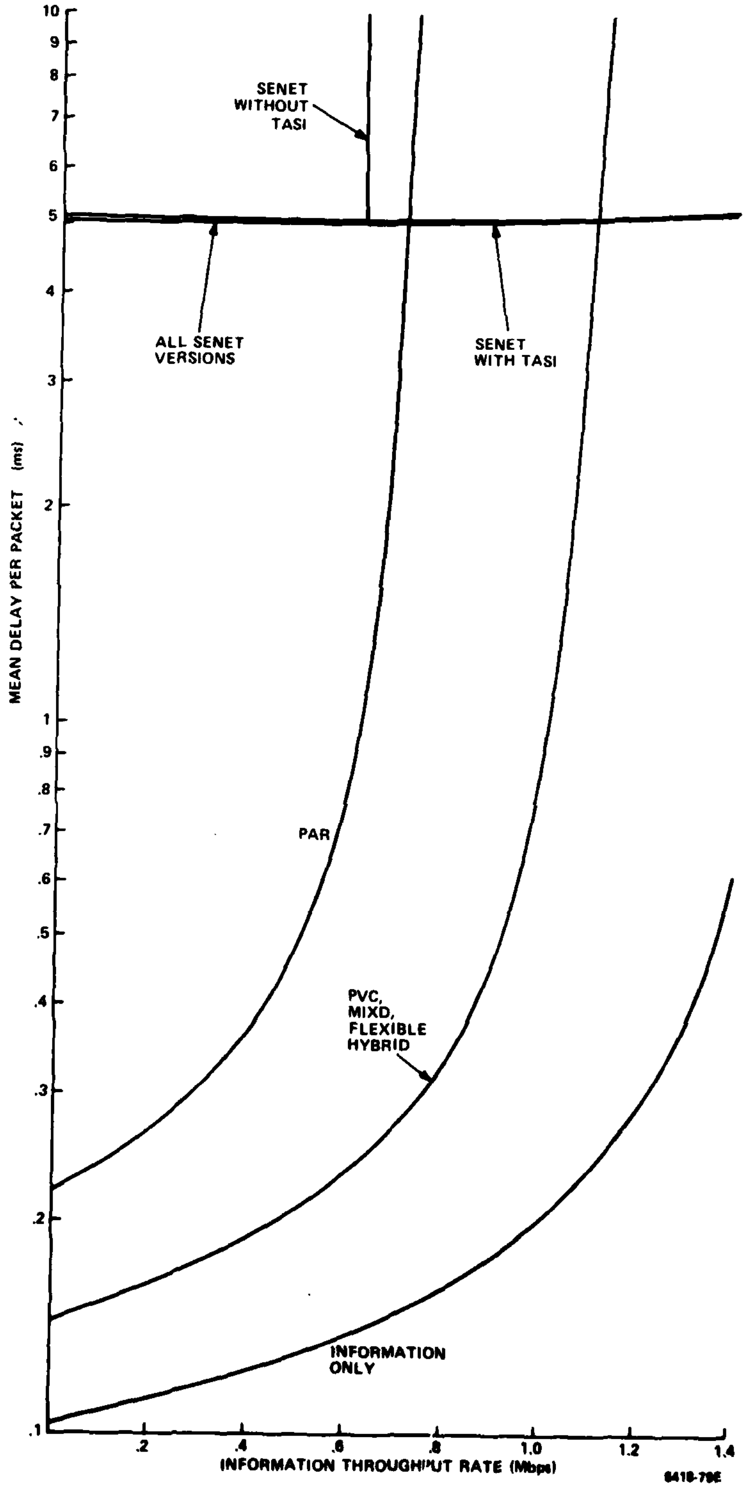


Figure 3.2-15. Delay-Throughput Relationship for a T1 Carrier Transmitting Pure Voice

therefore, be given by $\frac{t+T}{2}$. Frames arrive at the output trunk synchronously at a rate of one each t seconds. Each frame has at most one voice parcel per active voice call; therefore, each TDM slot can be modeled as a D/D/1 server, with a maximum arrival rate of one voice parcel each T seconds. The actual rate, however, depends on whether or not TASI is applied. A voice activity factor of 0.4 is assumed here, and the trunk delay performance for both SENET and enhanced SENET are shown.

Figure 3.2-16 shows the delay-throughput performance for a T1 carrier transmitting pure interactive data. For this case, the SENET with variable frame, MIXD, and Flexible Hybrid schemes reduce to the PAR scheme. Except at very low throughput rates, these systems show superior performance over both the SENET with fixed frame and the PVC techniques.

Finally, Figure 3.2-17 provides the delay performance of bulk data under the different integration schemes. For the low and mid throughput range, PVC shows the best delay for bulk data over all other schemes. In the high throughput range, the Flexible Hybrid provides smaller mean delay for bulk data.

The above discussion relates to the performance of a single traffic type under the different integration techniques. The traffic in any network generally consists of a combination of all three types. Figure 3.2-18 shows the triangle that bounds all different mixture ratios of the three traffic types. Figures 3.2-19a and b display the recommended integration technique for the different information mixing ratios within the traffic triangle. Based on the above results, the following guidelines are recommended. Under low information throughput, either the PVC or the MIXD scheme should be implemented in the switches. Under high information throughput conditions, enhanced SENET and Flexible Hybrid are the best concepts from the viewpoint of delay versus throughput. Determination of the exact location of the boundaries between the areas of recommended application of the different concepts will be an important objective during test and evaluation of the terrestrial subnetwork TSN.

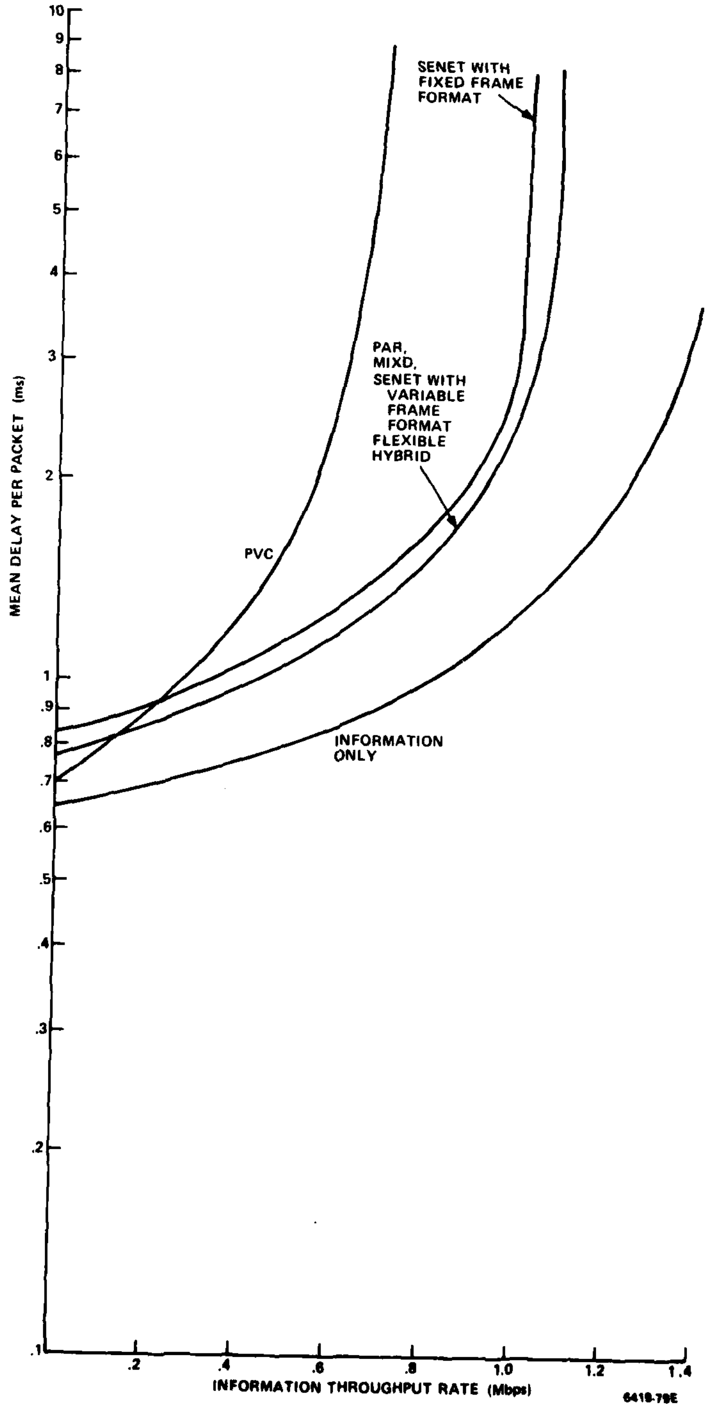


Figure 3.2-16. Delay-Throughput Relationship for a T1 Carrier Transmitting Pure Interactive Data

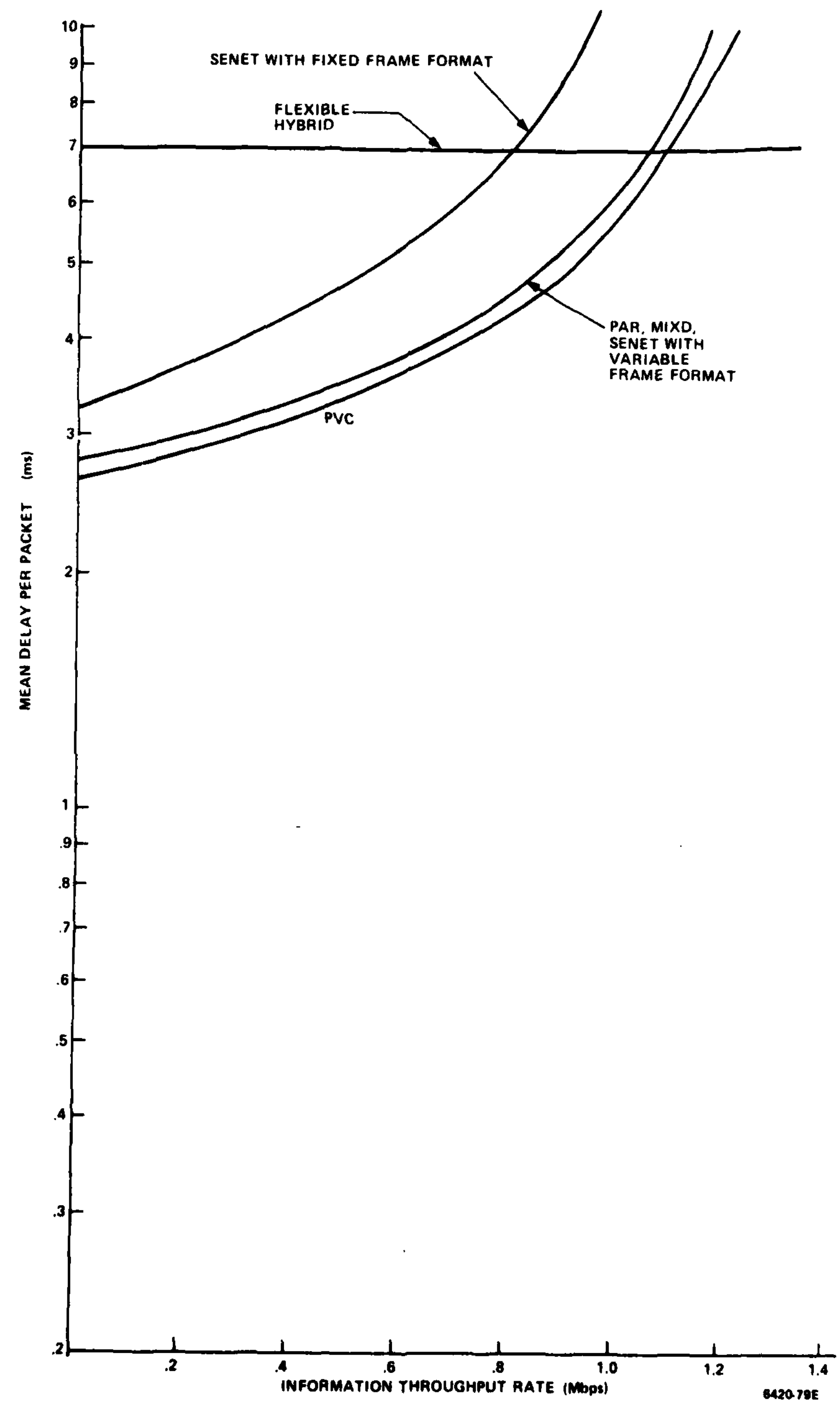


Figure 3.2-17. Delay-Throughput Relationship for a T1 Carrier Transmitting Pure Bulk Type Data

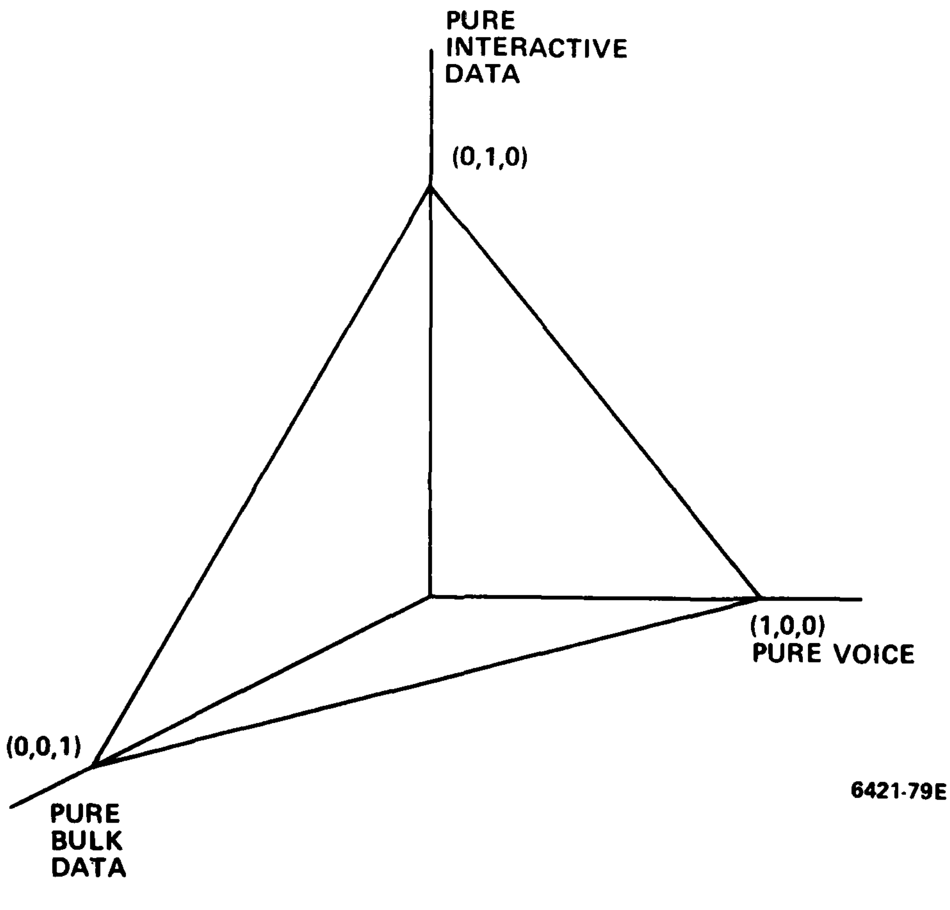
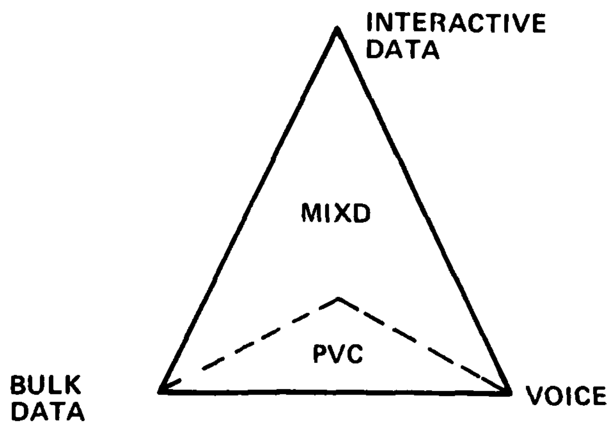
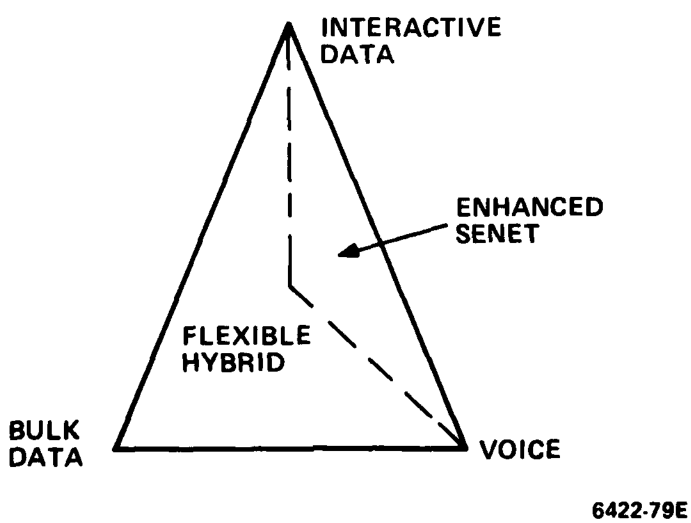


Figure 3.2-18. Traffic Mixture Triangle



(a) Low Information Throughput Rates



(b) High Information Throughput Rates

Figure 3.2-19. Guidelines for Recommended Integration Techniques

3.2.3.3 TASI and TADI

In order to make maximum use of system bandwidth, the use of tandem Time Assignment Speech Interpolation (TASI) and Time Assignment Data Interpolation (TADI) techniques to use the silent periods during voice conversation were investigated. Application of tandem TADI and TASI to voice consists of two operations. The first is the detection of silent or idle time intervals within the speech waveform. The second interpolation of voice or data within the idle periods, resulting in a two-fold improvement in transmission capacity for a large number of channels. Packet switching lends itself very well to tandem TASI and TADI techniques. During speech silent intervals, packet transmission ceases and the available capacity can be used to transmit other data. This helps contribute to the high transmission efficiency exhibited with packet switched voice. Unfortunately, the complexity of the switch to perform speech and silence detection is increased; however, since transmission costs generally exceed switch costs, this results in an overall reduction in total costs for the network.

These techniques can be applied equally as well to a hybrid system, wherein voice is circuit switched. One of the problems of applying tandem TASI to a SENET-like approach is that the dynamic adjustment of link maps following a call termination results in an additional complexity needed to coordinate changes in these maps. Transmitting and receiving switches must agree exactly on what is located where, how much of each master frame each allocation requires, and the exact timing of constantly changing allocation maps. This results in an increased coordination complexity that hampers the interpolation of speech into the silent periods of other speech.

Several methods were proposed for achieving tandem TASI in a hybrid system. Active/silent status from CCIS messages propagated from other nodes may be used to determine imminent changes in slot utilization; however, CCIS message transmission is not synchronized with the trunk channel. CCIS messages occur one frame (or more) later than detection of active/silent status of a subscriber. Late or out-of-sequence messages about one user's transmission status could disrupt all other users in the system. Status may also be derived

from local detection of silence bit patterns on the incoming and outgoing trunk channels correlated with the resident "connect map." One problem with this method is that if the voice is encrypted at the source, then silence detection cannot be carried out at each intermediate node. A third method is to include a control vector in the circuit switch portion of the frame that would identify those talkers who are active during that particular frame; however, as implied in the last paragraph, the requirements for rapid manipulation of frame maps and node-to-node forwarding of control vectors would make the achievement of tandem TASI via this approach very difficult.

All of the approaches considered thus far either have very tight time constraints or appear to have unsolvable problems associated with them. Contrast this with the ease in which tandem TASI is provided in a packet switching system. As a result of the analysis, a hybrid system concept using tandem TASI has not been recommended for inclusion in the TSN at this time; however, a means for implementing circuit-like switched voice is provided by the synchronous portion of the frame in the Flexible Hybrid concept. If it becomes desirable to do so during the experimental phase, the application of tandem TASI to the hybrid model could be reconsidered at that time.

The interpolation of packet switched data into the silent periods of voice using TADI appears to be practical. A control vector in the circuit portion of the frame would provide active/silence status for subscribers during that frame. The switch would use the control vector to determine which allocations in that particular frame are handled as normal voice calls, and identify the silent allocations into which data packets would be interpolated. It is possible to eliminate the separate circuit and packet switch regions altogether, allowing the circuit switched data to fill most of the master frame except for the start of frame marker, the control vector and a minimum CCIS region in order to get a further increase in efficiency.

The concept evaluation study showed that inclusion of TADI in the hybrid system results in an increase in efficiency of 10 to 15% for high voice/data mixtures. The increase does not result in an efficiency that compares favorably with either hybrid switching using

TASI techniques or packet switching systems; therefore, a hybrid system concept using TADI has not been recommended for the TSN. However, like TASI, it can be included at a later date if circumstances warrant.

3.2.3.4 Circuit Reservation and Allocation

The objective of circuit reservation and allocation is to minimize the time interval between capacity dedication and transmission for circuit switched calls in a hybrid system. This is effectively accomplished in a packet switching system through the PVC method in that capacity is never allocated until actually used; however, the PVC path traced by constituent packets is reserved beforehand. The results indicate that circuit reservation and allocation provide about a 2 to 6% improvement in the transmission efficiency of the hybrid system; therefore, because it provides a way to squeeze the maximum amount of efficiency out of a hybrid system, we plan to include circuit reservation and allocation for the circuit switched data in the Flexible Hybrid model (see Section 4.3).

3.2.3.5 Packet Lengths

Packet length has a significant impact on transmission efficiency with regard to the overhead associated with packetized voice and various data transmissions. Transmission efficiency is directly affected by data packet length. For small packet sizes the ratio of overhead to information bits is high causing the efficiency to decrease. At large packet sizes this overhead is reduced; however, the retransmission of erroneous packets comes into effect. Depending on the bit error rate, the effect can again be a reduction in transmission efficiency.

These results depend greatly on the number of header bits required for each data packet. During the concept evaluation study, data packet length was examined over a range of 250 to 2000 bits and the data header over a range of 50 to 250 bits. When the data header was large, the curves for transmission efficiency were lower and fell

off more rapidly at the smaller data packet sizes than the corresponding curves for smaller data headers. A typical data packet length would be 1000 bits with a header length of 150 bits. This provides a transmission efficiency on the flat portion of the curves with an overhead typical to that found in the AUTODIN II network.

Transmission efficiency is also a function of voice packet and header length. The voice packet consists of a number of bits accumulated during a fixed accumulation period, (e.g., 1000 bits during a 67 ms interval for a 16Kbps rate, and 160 bits for a 2400 bps rate), or the number of bits accumulated during several such intervals, (e.g., 960 bits accumulated during a 402 ms interval for a 2400 bps rate). In general, the smaller the packet size, the lower the overhead efficiency. Likewise, the larger the number of overhead bits, the smaller the overhead efficiency. For PAR voice traffic, the header was taken as 150 bits, the same as that of data packets. As expected, this method was less efficient than the other two packet switching concepts for any of the vocoder rates examined (2, 4, 4 and 16 kbps). In the PVC method, the headers for both voice and data packets are only 32 bits; so one expects the overhead efficiences to be high, and this was shown to be the case. The PVC method of transmission implies: elimination of the check field; sending address for voice; and storage of repetitive information, such as call identification and routing at the intermediate nodes. This results in a simplified header, hence, the reduction in overhead bits from 150 to 32. A disadvantage is the loss of error protection and capability for alternate routing around failed paths. The MIXD concept uses PVC routing for voice, with a 32-bit header, while the interactive data traffic is sent PAR with a 150-bit header. This concept, as expected, falls between the other two in overhead efficiency, since some traffic is sent with one mechanism and some with the other; however, it performed better than the all-PVC method in terms of throughput efficiency.

An extension of the PVC voice strategy is to group together all the packets from the established voice circuits of a particular link into a "super" packet, which is transmitted using a single header.[16] This modification was not studied during the concept evaluation; however, it can potentially reduce the overhead to nearly that of circuit switching while still retaining most of the flexibility and effectiveness of packet switching. It also adds to the delay and complexity because of the tandem TASI, like functions required to gather the "super-packet" together at one end of the link and separate out the individual packets at the other end.

3.2.3.6 Traffic Handling

Various methods for handling the different classes of traffic were investigated during the concept evaluation. Table 3.2-1 shows the traffic handling technique recommended for each type of data for those integrated voice/data concepts that showed the most favorable characteristics in terms of transmission efficiency, delay, etc. It also shows the most viable candidates for future experimentation. In both the MIXD and Flexible Hybrid systems, voice will be transmitted via PVC routing techniques; however, FAX, burst and bulk data will be synchronously switched in the Flexible Hybrid system while PVC will be used in the MIXD system.

Interactive, query/response, narrative/record, and some control messages will be adaptively routed under both concepts. The assignment of each traffic type to a specific traffic handling technique plays an important part in the high efficiency obtained for these particular concepts.

Precedence/priority/preemption are strategies used for controlling the adjustment of synchronous and nonsynchronous traffic in each of the concepts to provide service to each class to traffic. This ensures that adequate capacity is allocated to provide an acceptable grade of service and connection delay for voice and facsimile calls, a satisfactory acceptance and delivery delay for data traffic, and the achievement of high efficiency on the links. In the

TABLE 3.2-1. CHART OF TRAFFIC HANDLING FOR VARIOUS DATA CLASSES

Application Type			MIXD Packet		Flexible Hybrid		
Class	Delay	Subclass	PVC	PAR	Synch	PVC	PAR
Real Time	Very Low	Voice	X			X	
		FAX	X		X		
Non-Real Time	Low	Interactive		X			X
		Query/Response		X			X
		Narrative/Record		X			X
	High	Bulk 1	X		X		
		Bulk 2	X		X		
Burst	Low	Wideband Information Bursts	X		X	*	
Control	Low	Trailblazer Packets CCIS, PAR Control, etc.	*	X		*	X

*Candidate for future experimentation.

Flexible Hybrid system, each traffic class (i.e., circuit or packet switched traffic) exerts an effect on the other class. A movable boundary arrangement between classes provides a more efficient use of transmission capacity than a fixed boundary, since one class of traffic can utilize idle traffic capacity normally assigned to the other. It yields better packet delay performance than a fixed boundary and exhibits less sensitivity to changes in the circuit switched traffic blocking probability.[14] Another technique to increase the transmision efficiency and at the same time decrease the mean packet delay in the Flexible Hybrid is to allow the frame to have a variable length.[12] A packet transmission near the end of the frame would not be suppressed, or interrupted, in order to transmit a new master frame, but would be completed. Under these conditions, the transmission delay of circuit switched calls is no longer constant, but exhibits jitter as a result of the variable frame; however, since the circuit switched region is used for bulk data, facsimile, and bursts, the additional delay will not have a significant impact on delivery performance for these types of traffic.

3.2.3.7 Conclusions

The concept evaluation study showed that packet switching had many advantages for integrating voice and data in terms of efficiency, delay, and required complexity. The MIXD routing strategy, a combination of the virtual circuit and adaptive routing approaches, attempts to associate each routing strategy with each class of for which it is best suited; hence, voice packets use a fixed path protocol while data packets use a path independent protocol. Although more complex than the other packet switching concepts, it is recommended here because of its performance, and because of the need to treat voice and data packets in a separate fashion. The latter consideration is the result of the effect delay and variability of delay have on the intelligibility and quality of speech communications.[15] Since the speech data stream has stringent delay requirements, a network designed to handle packetized speech will be required to prioritize voice packets over data packets to limit the path length and minimize the effect of delays. When data traffic is serviced, the data packets are provided with full

headers and follow a path determined by adaptively updated routing tables found at each node. In this scheme, noninteractive data may be sent using the protocol that is most appropriate, (i.e., PVC for bulk data, PAR for others).

One advantage of the MIXD approach is the capability of providing an all-virtual or all-adaptive routing of voice and data because of the commonality among the PVC, PAR, and MIXD concepts. It should be noted that provision of this capability will require not only changes in the routing algorithms, but also changes in the protocol and flow control procedures as well; however, it will allow a maximum flexibility in the experimental test bed to test the key variations within the generic packet switching family. This will be addressed in later sections.

Another concept recommended for the TSN network is the Flexible Hybrid scheme. Here, voice and interactive data are handled in a packet switched mode, as in the MIXD system, and bulk data is handled in a circuit switched mode. The concept evaluation study showed high transmission efficiency for the Flexible Hybrid approach, although the number of functions required to implement this concept is expected to be higher than that of the MIXD packet switching system. A combination hybrid switching scheme had been suggested earlier by various investigators as a very cost-effective strategy by which to integrate voice and data into common system.[8,9] The advantages stem from the high efficiency obtained by packet switching of voice and interactive data, and the reduced switching and channel capacity requirements when circuit switching bulk data messages. Although the efficiency gained in handling bulk data in the circuit region rather than as PVC packets is small, the advantages of lower processor overhead and the effects of having a synchronous region in a hybrid switch justify the experimental concept. This presumes a simplex tranmission for bulk data with error control provided either via packets or a small transmission capacity allocated in the reverse direction. One hypothesis to be evaluated experimentally is whether the improvement in efficiency obtained by transmitting bulk data in the synchronous region is more than offset by the increased complexity

necessary to implement two kinds of switching, (i.e., circuit and packet), in this system. In addition, measured performance in transmitting bulk data in this system must be compared to the use of PVC to transmit bulk data in a packet switching system.

Although a circuit switching concept is not recommended in the experimental test bed, a circuit switching technique is inherent in the hybrid approach for bulk data transmission. The Flexible Hybrid system attempts to combine voice and data in a way that matches the traffic characteristics of a particular type of data to the switching concept that optimizes its transmission. Thus, the synchronous transmission of long bulk data messages seeks to take advantage of the low delay and high connectivity associated with circuit switching techniques; therefore, although pure circuit switching as a generic concept is not a suitable candidate for the switching of both voice and data, it is present as a subset of the Flexible Hybrid system.

3.3 ADVANCED INTEGRATED SYSTEM CONCEPT ARCHITECTURE

This subsection describes the architectures of the packet switching and hybrid switching approaches which appeared as the most promising advanced system concepts to be implemented in a test bed. Paragraph 3.3.1 addresses functional requirements for each approach. Paragraph 3.3.2 discusses functional architecture for each of the concepts recommended for implementation. Paragraph 3.3.3 then discusses the various operational strategies as they would be incorporated into the individual switching approaches.

3.3.1 Functional Requirements

A functional analysis has been performed in which each of the switching concept variations has been analyzed in terms of the various functions that would have to be included in an implementation of the concept. The functions identified are believed to be architecture-independent and are subject to hardware/firmware/software tradeoffs. In addition, a number of functions may well be combined in an implementation. Although no attempt has been made in apply weighting

to the functions, we have tried to break down complex functions into their constituent (and simpler) subfunctions where possible. In all, over 80 functions have been identified, falling into eight categories:

1. Protocol Functions
2. Input/Output (I/O) Handling Functions
3. Routing and Circuit Setup/Teardown Functions
4. Packetized Voice Handling Functions
5. Precedence, Priority, and Preemption Functions
6. Frame Maintenance Functions
7. TASI Functions
8. TADI/ADM Functions.

Each category of function is described below. Table 3.3-1 summarizes the numbers of functions in each category required to implement the concepts investigated.

3.3.1.1 Protocol Functions

This category of functions handles the various link-level and source/destination-transport-level protocols required for the switching concepts under consideration.

a. Link-level voice protocol
b. Link-level data protocol
c. Variable-path voice protocol
d. Variable-path data protocol
e. Fixed-path voice protocol
f. Fixed-path data protocol
g. Synchronous-voice CCIS protocol
h. Synchronous-data CCIS protocol
i. Synchronous-data "fast" protocol.

3.3.1.2 Input/Output Handling Functions

The task of these functions is only the handling of input and output traffic, and does not include subsequent processing of input data or preliminary processing of output traffic. The following functions have been identified:

a. Data-packet input handling
b. Voice-packet input handling
c. Data-packet output handling
d. Voice-packet output handling
e. Synchronous-data input handling
f. Synchronous-voice input handling
g. Synchronous-data output handling

TABLE 3.3-1. FUNCTIONAL SUMMARY

FUNCTION \ CONCEPT	CIRCUIT				PACKET			HYBRID					
	TRAD	FAST ③	ENH ①	ENH ②	PVC	PAD	MIXD	GENET	ENH ①	ENH ②	VL ②	HC ②	FLEX
SUMMARY													
CATEGORY	NUMBER OF FUNCTIONS												
PROTOCOL	2	3	2	2	4	4	5	4	4	4	4	4	5
I/O HANDLING	6	6	6	6	4	4	4	9	9	9	8	10	8
ROUTING & CKT SETUP	5	6	5	5	7	4	7	9	9	9	9	10	10
PACKETIZED VOICE HANDLING	—	—	—	—	3	4	3	—	—	—	—	—	3
PREC., PRIORITY, PREMPT.	3	3	3	3	6	5	6	5	5	5	5	6	7
FRAME MAINTENANCE	2	2	2	2	—	—	—	6	6	6	7	4	7
TASI	—	—	7	11	2	2	2	—	8	12	12	13	2
TADI/ADM	—	—	6	9	2	2	2	2	7	10	10	10	2
TOTAL	18	20	31	38	28	25	29	35	48	55	55	57	44

5555-79E

① ASSUMES LINK-BY-LINK TASI/TADI/ADM

② ASSUMES TANDEM TASI/TADI/ADM

③ NOT IMPLEMENTABLE WITH CURRENT STATE-OF-THE-ART TECHNOLOGY

h. Synchronous-voice output handling
i. CCIS input handling
j. CCIS output handling
k. Partial packet handling
l. Noncontinguous input demultiplexing.

3.3.1.3 Routing and Circuit Setup/Teardown Functions

This category of functions handles tasks associated with the routing of various kinds of traffic, the establishment and release of the data structures associated with various kinds of circuits, and the assignment of transmission bandwidth in various types of calls. The following functions have been identified:

a. Adaptive routing
b. Routing-table updating
c. Virtual-circuit routing
d. Deterministic alternate routing
e. Real-circuit routing
f. Variable-bandwidth reservation
g. Variable-bandwidth allocation
h. Fixed-bandwidth allocation
i. "Best fit" channel allocation
j. Simplex real-data circuit setup/teardown
k. Simplex virtual-data circuit setup/teardown
l. Duplex real-data circuit setup/teardown
m. Duplex real-voice circuit setup/teardown
n. Duplex virtual-data circuit setup/teardown
o. Duplex virtual-voice circuit setup/teardown
p. Virtual-data connection setup/teardown
q. Virtual-voice connection setup/teardown
r. "Fast" circuit setup/teardown.

3.3.1.4 Packetized Voice Handling Functions

These functions are concerned with the unique requirements of packetized voice handling:

a. Voice Packetizing
b. Voice Depacketizing
c. Voice Silence Detection (also appears as a TASI function)
d. Voice Activity Detection (also appears as a TASI function)
e. Voice reassembly
f. Voice gap control.

3.3.1.5 Precedence, Priority, and Preemption Functions

This category of functions handles the precedence, priority, and preemption tasks as follows:

a. Precedence schemes reconciliation
b. Transmission priority mapping
c. Priority queue management
d. Virtual-circuit bandwidth preemption
e. Real-circuit bandwidth preemption
f. "Best fit" preemption
g. Voice-packet discard
h. Switch resource preemption.

3.3..1.6 Frame Maintenance Functions

This group of functions is peculiar to the circuit switching and hybrid concepts inasmuch as the packet switching concepts are asynchronous. They are concerned with the synchronization and maintenance of the frame structure.

a. Start-of-frame generation
b. Start-of-frame detection
c. Variable-length frame management
d. Frame padding
e. Class I/Class II boundary detection
f. Class I map maintenance
g. Class I compaction

3.3.1.7 TASI Functions

This group includes functions required for implementation of TASI in the circuit switching and hybrid switching concepts. TASI is considered to be inherent in the packet switching concepts and requires the detection of voice silences and activity as indicated below. Two types of TASI have been considered for the circuit and hybrid concepts: link-by-link TASI and end-to-end (or tandem) TASI. The link-by-link TASI functions are a subset of the tandem TASI functions when considered at the level of this report, but would probably require separate implementations at the design and build levels. Currently identified functions are:

a. CCIS TASI protocol
b. Silence-code insertion
c. Active/Inactive TASI-channel mapping
d. TASI "operating line" management

e. TASI-channel selection
f. "Best fit" TASI mapping
g. Voice-silence detection
h. Voice-activity detection
i. TASI-map compaction
j. Network Talk-Spurt signaling
k. Network Talk-Spurt routing
l. Network Talk-Spurt synchronization
m. Network Talk-Spurt gap control.

3.3.1.8 TADI/ADM Functions

This group includes functions required for the implementation of ADM in the circuit switching concepts and TADI in the hybrid switching concepts. TADI is considered to be inherent in the packet switching concepts and requires the detection of data silences and activity. As in the case of TASI, two kinds of TADI/ADM are considered, link-by-link and end-to-end. The functions identified below are the approximate equivalents of the corresponding TASI functions:

a. CCIS TADI/ADM Protocol
b. Active/Inactive TADI/ADM Channel Mapping
c. TADI/ADM "Operating Line" Management
d. TADI/ADM Channel Selection
e. Data-Silence Detection
f. Data-Activity Detection
g. TADI Map Compaction
h. Network Data-Spurt Signaling
i. Network Data-Spurt Routing
j. Network Data-Spurt Synchronization.

3.3.2 Functional Architecture

This paragraph describes the functional architecture of two of the most promising switching concepts: the MIXD Packet Switching Model and the Flexible Hybrid Switching Model. In addition, the architectures for three optional switching models are also discussed.

3.3.2.1 MIXD Packet Switching Model

Under the packet switching concept, both voice and data are packetized, sent through the network, and reassembled into complete messages or voice streams at the receiver; however, different packet sizes, formats, and protocols may be used for speech and data. The PVC concept is a more efficient method for voice packet transmission,

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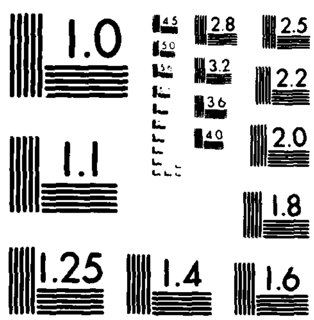

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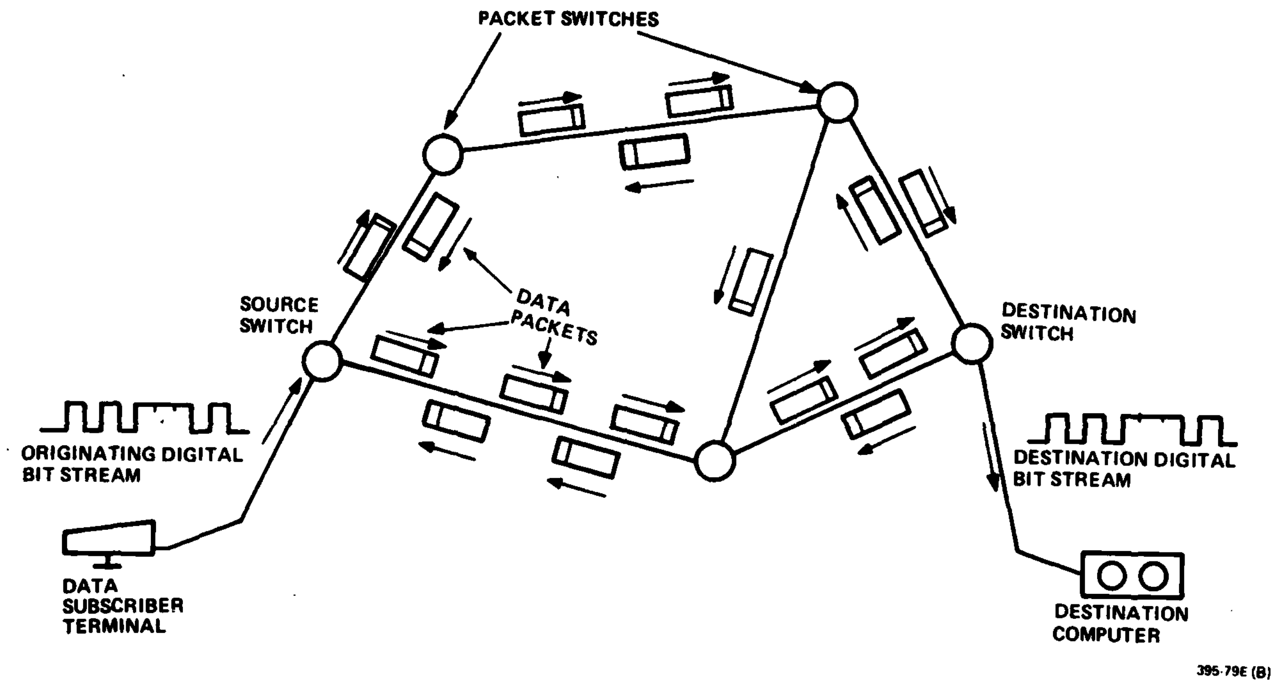
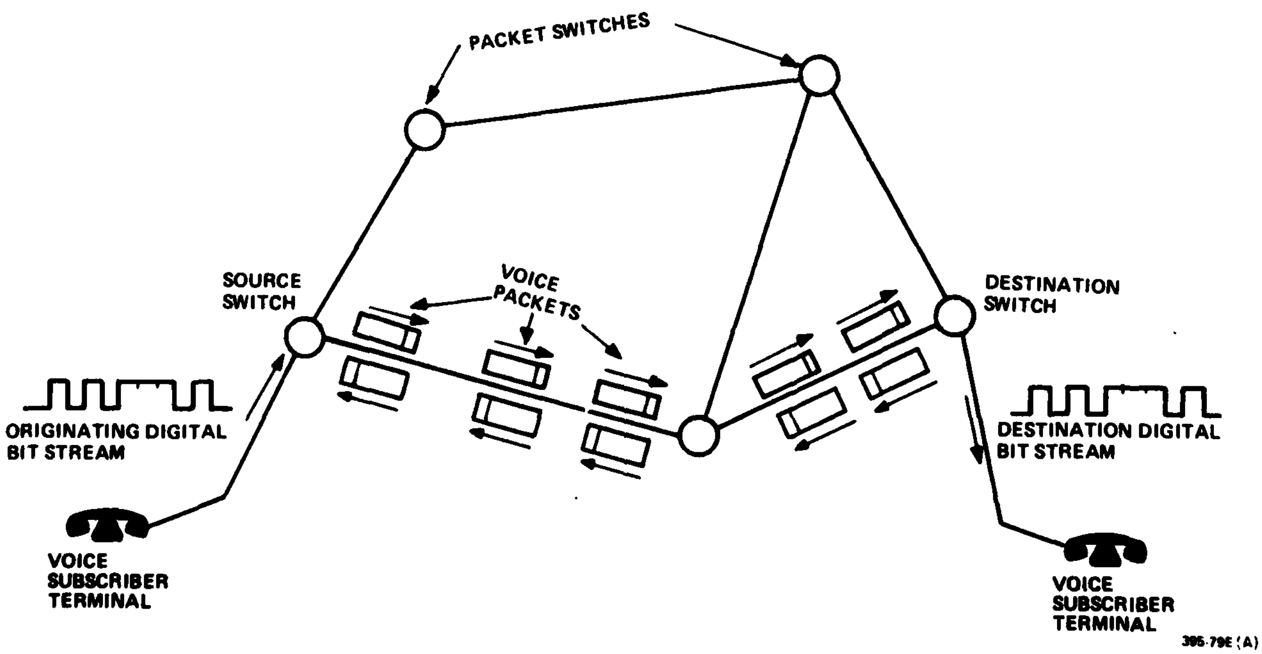
while the PAR concept would be more efficient for data packet transmission. The packet switching model chosen is based on a strategy in which the voice packets are routed over virtual paths as in the PVC scheme, (see Figure 3.3-1a) and data packets are independently routed as in the PAR scheme, (see Figure 3.3-1b) to avoid congestion points. This approach, called MIXD, is a combination of the Packet Virtual Circuit (PVC) and Packet Adaptive Routing (PAR) approaches, and attempts to associate each routing strategy with the class of data for which it is best suited; hence, voice packets use the fixed path protocol while data packets use the adaptive protocol. To transmit each packet under control of the strategy that is best suited to its characteristics, two separate transport protocols are required. It will be necessary to treat voice and data separately due to different transmission requirements. For example, voice continuity is sensitive to the delay variations of the speech packets of a specific conversation; therefore, to minimize these variations, a higher priority of voice over data packets and a uniform transmission path for voice packets will be required.

The MIXD concept transmits voice traffic and data traffic in packetized form, but treats the voice packets and data packets with different protocols that are tailored to the special characteristics of each kind of traffic. When a voice subscriber requests a connection, a fixed path protocol establishes a virtual circuit. The protocol considers the available bandwidth on each trunk when making the routing decisions. The requesting subscriber specifies an initial bandwidth, and this is used in determining capacity requirement. If no path can be found in which all links have sufficient available capacity, then the connection is blocked and the subscriber notified. The connection may have the capability to preempt bandwidth; e.g., a military network would have priority controls to give better service to certain voice subscribers (lower blocking) and to certain data subscribers (lower delay).

Once the virtual circuit for voice type transmission has been established by dispatching a trailblazer packet and receiving an affirmative response, the information path is in operation for the duration of the conversation. The subsequent voice packets carry



(a) Packet Virtual Circuit (PVC) Voice



(b) Packet Adaptively Routed Data

Figure 3.3-1. Packet-Switched Information Flow

minimal header information and are routed along the fixed routes established by the trailblazer. The voice packets are given priority on the trunks over the data packets within the constraints of the traffic prececdence scheme. The virtual circuit is disestablished when a caller hangs up.

When data traffic is presented for transmission, the data packets are furnished with complete headers, any required connections are established, and the data packets follow a path determined by adaptively updated routing tables in the nodes. The update operation is performed periodically in each node. The data packet is assigned by the routing table to a trunk queue that is managed by priority schemes in each node. The trunk choice is based on both the destination address and the network connectivity and congestion. As these change, different packets to the same destination from the same source may follow different paths. The data packets are subject to flow control restrictions; the voice packets, however, may be discarded up to a certain percentage. Voice packets will be discarded before data packets if they both have the same precedence.

The MIXD concept architecture is shown in Figure 3.3-2, which identifies the functions required to handle the various types of traffic.

Figure 3.3-3 shows the packet switched data flow at a MIXD switch. Incoming packets are first stored in memory through a packet storage process. At the same time, a pointer to the stored packet is placed in an input processing queue. The received packets are then subsequently processed in the order of their arrival and routed by the appropriate algorithm based on traffic type. In the MIXD concept, voice and bulk data packets are routed over a virtual circuit by a fixed path routing algorithm, whereas interactive data packets are routed independently by an adaptive routing algorithm. The routing algorithms place the stored packet pointers in the appropriate output queue; the packets themselves remain in memory where they were stored. Following the examination of pointers in the output queue, packets are extracted from memory and either dispatched over an outgoing trunk to the next switch or delivered to a locally terminated packet switched subscriber.

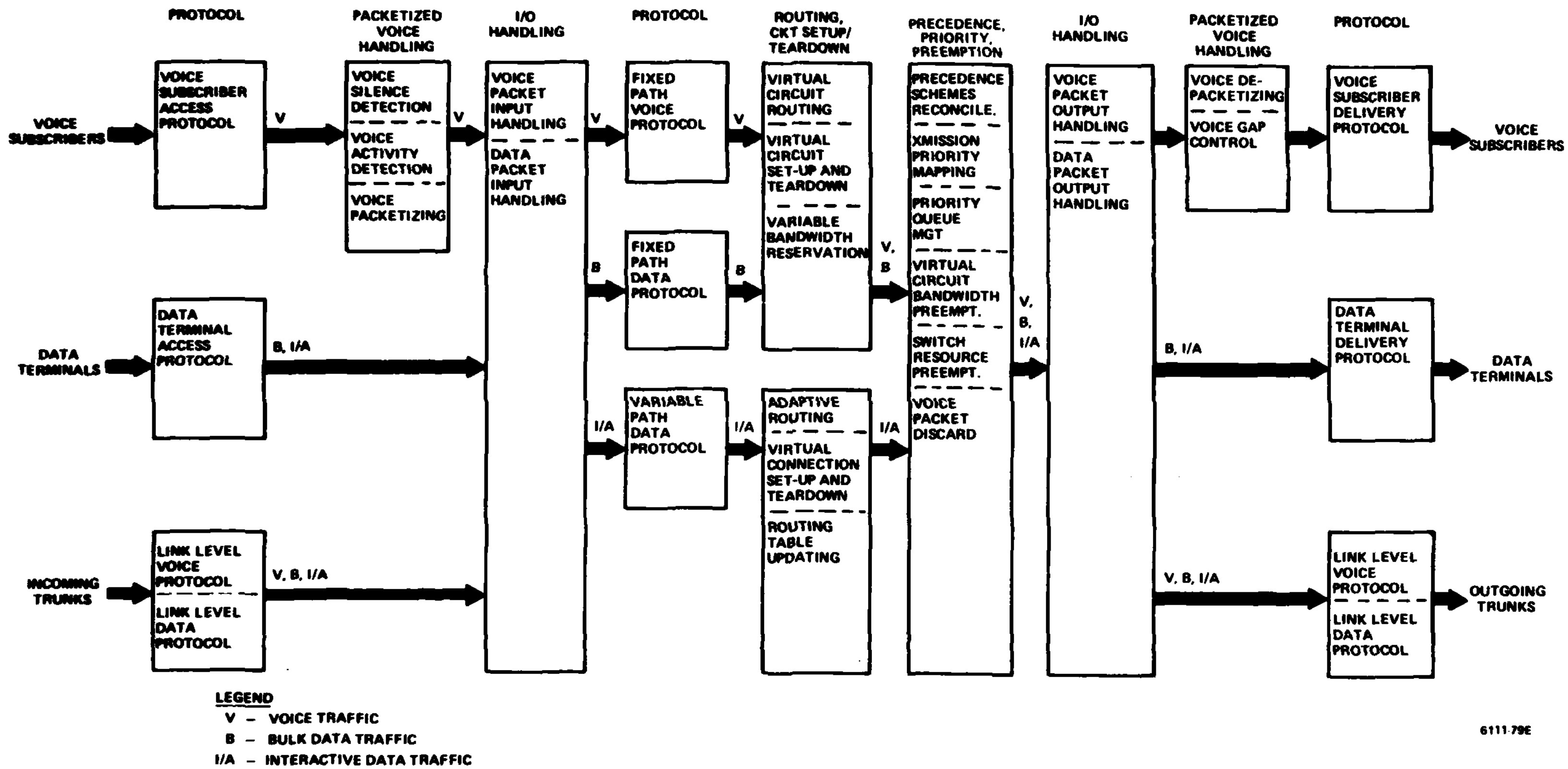


Figure 3.3-2. Mixed Concept Architecture

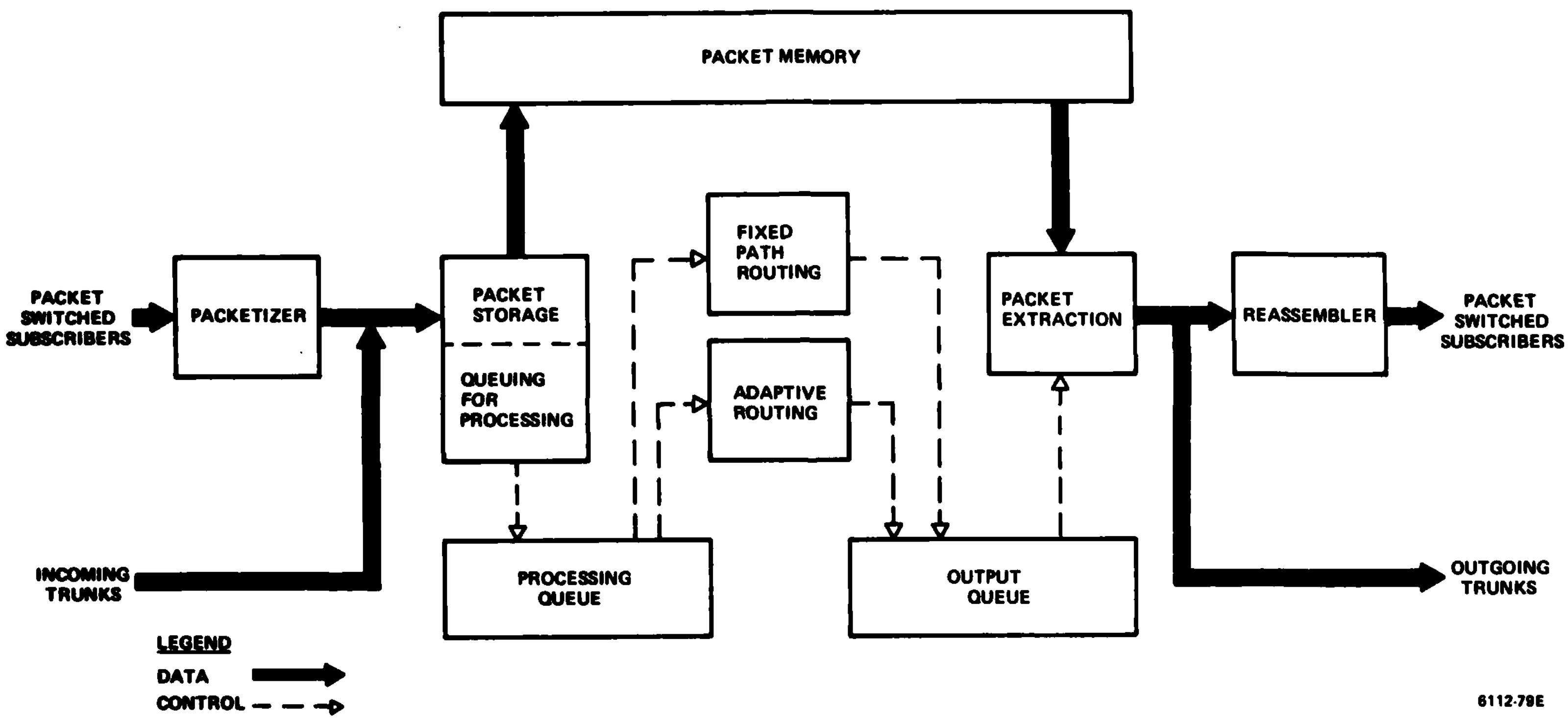


Figure 3.3-3. Packet-Switched Data Flow at a MIXD Switch

3.3.2.2 Flexible Hybrid Switching Model

The concept evaluation study shows that transmission efficiency of hybrid systems suffers in comparison to packet systems as a direct consequence of the inefficiency in transmitting circuit switched voice. TASI is one method to improve the efficiency, but at the price of adding a considerable amount of complexity. The Flexible Hybrid concept for integrating voice and data into a common system is an approach that would help to alleviate this condition. It illustrates that the different integrated voice/data concepts are not necessarily discrete and exclusive, but rather are entries on a continuum of techniques in which performance and control blend from one technique to another.

The Flexible Hybrid scheme is shown in Figure 3.3-4. Wideband digital trunk capacity is divided into contiguous master time frames. The frame period, for example, 10 or 20 ms, once selected, is the same throughout the network. Within a frame, portions are used as:

1. A marker to indicate the start of a frame;
2. A region that multiplexes circuit switched synchronous traffic;
3. A region that incorporates packet switched or message switched traffic handled as packet switched data.

Voice is transmitted as packet switched data without error control via the PVC routing algorithm; interactive and narrative/record data are transmitted as error protected packet switched data; bulk and burst data information transmitted in simplex form as circuit switched data. This assumes either that no error control is required, or that error control is provided via packets or a small transmission capacity allocated in the reverse direction. There are several advantages to this method. Bulk data is transmitted more efficiently and with less delay using circuit switching since, in general, long messages favor circuit switched operations and short messages favor packet switched operations.[10] Packet switching of voice provides a TASI-like efficiency advantage without greatly increasing complexity. During speech

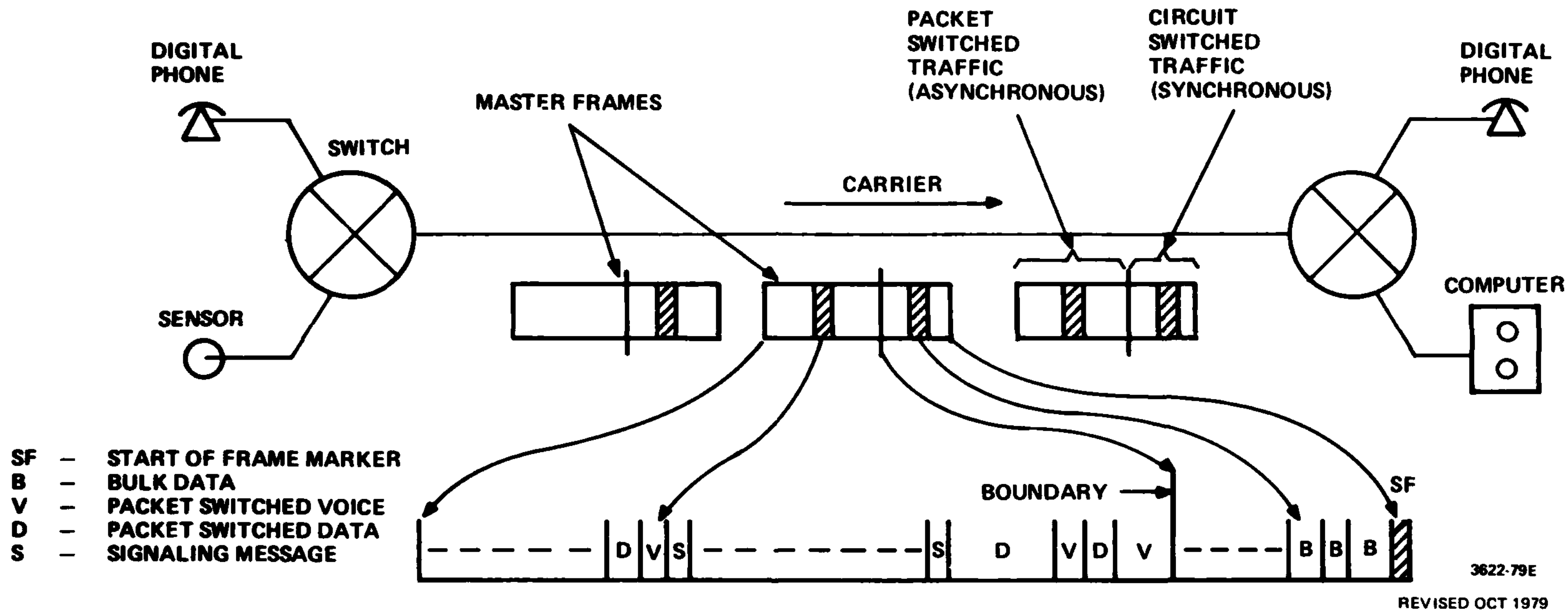


Figure 3.3-4. Flexible Hybrid Channel Capacity Structure

silences, transmission ceases; hence, the capacity is available for servicing other users. Packet switched traffic or message switched traffic requiring error protection is handled conveniently within the packet switched mode of operation.

The master frame concept employing a movable boundary is used to carry the circuit switched and packet switched regions. An improvement in overall transmission efficiency is close to 100% (assuming bulk data is handled in simplex form by the circuit switched region). Figure 3.2-13 shows this as a function of voice/data distribution. It is seen that for voice rates of 16Kbps, a theoretical efficiency of up to 98% is obtainable using this method.

The Flexible Hybrid concept architecture, shown in Figure 3.3-5, identifies the functions required to handle the various types of traffic.

Figure 3.3-6 illustrates the circuit and packet switched data flow at a Flexible Hybrid switch. The trunk channel is synchronously clocked, resulting in frames of fixed-time duration. Frames arriving on an incoming trunk are separated into circuit switched and packet switched information, which is then transferred appropriately to packet or circuit processing. Signals from circuit switched subscribers are transferred to circuit switch processing at constant time intervals. Here, a new circuit switched region is generated based on the buffered data from local circuit switched subscribers and the bits in the circuit switched region transmitted from the preceding node. Voice signals and messages from packet switched subscribers are digitized if necessary, formed into packets with an added header, and placed in the packet buffer queue along with packets that have arrived from the preceding node and require tandem transfer to a subsequent node.

At tandem switches, packet processing is kept to a minimum (consisting of a read-in, header analysis, and read-out). Packets arriving at the destination node are analyzed and the header removed. Reassembly of packets takes place if required, and the resulting bit stream is presented to the subscriber. An asynchronous/synchronous interface is required for voice packet operation, since packets are

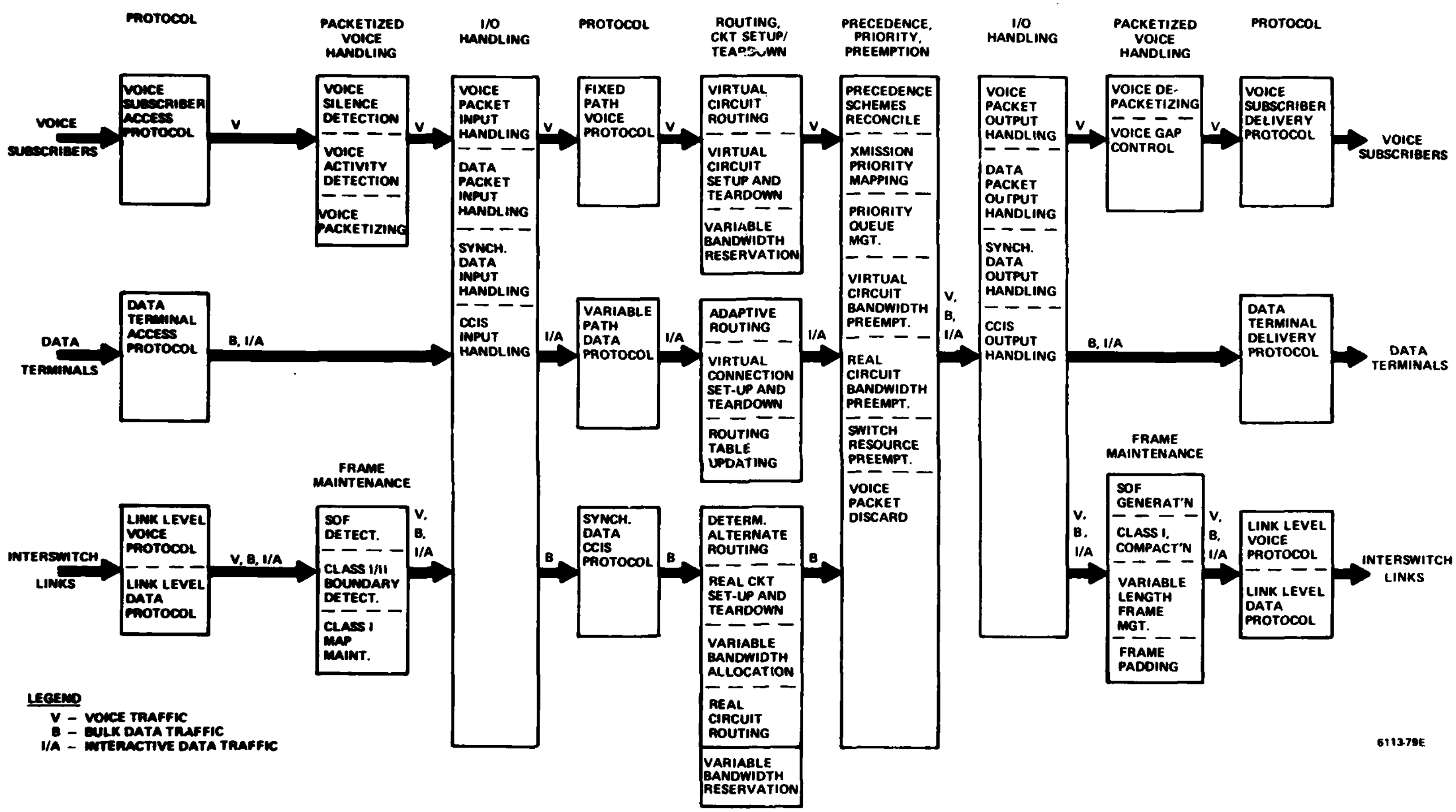


Figure 3.3-5. Flexible Hybrid Concept Architecture

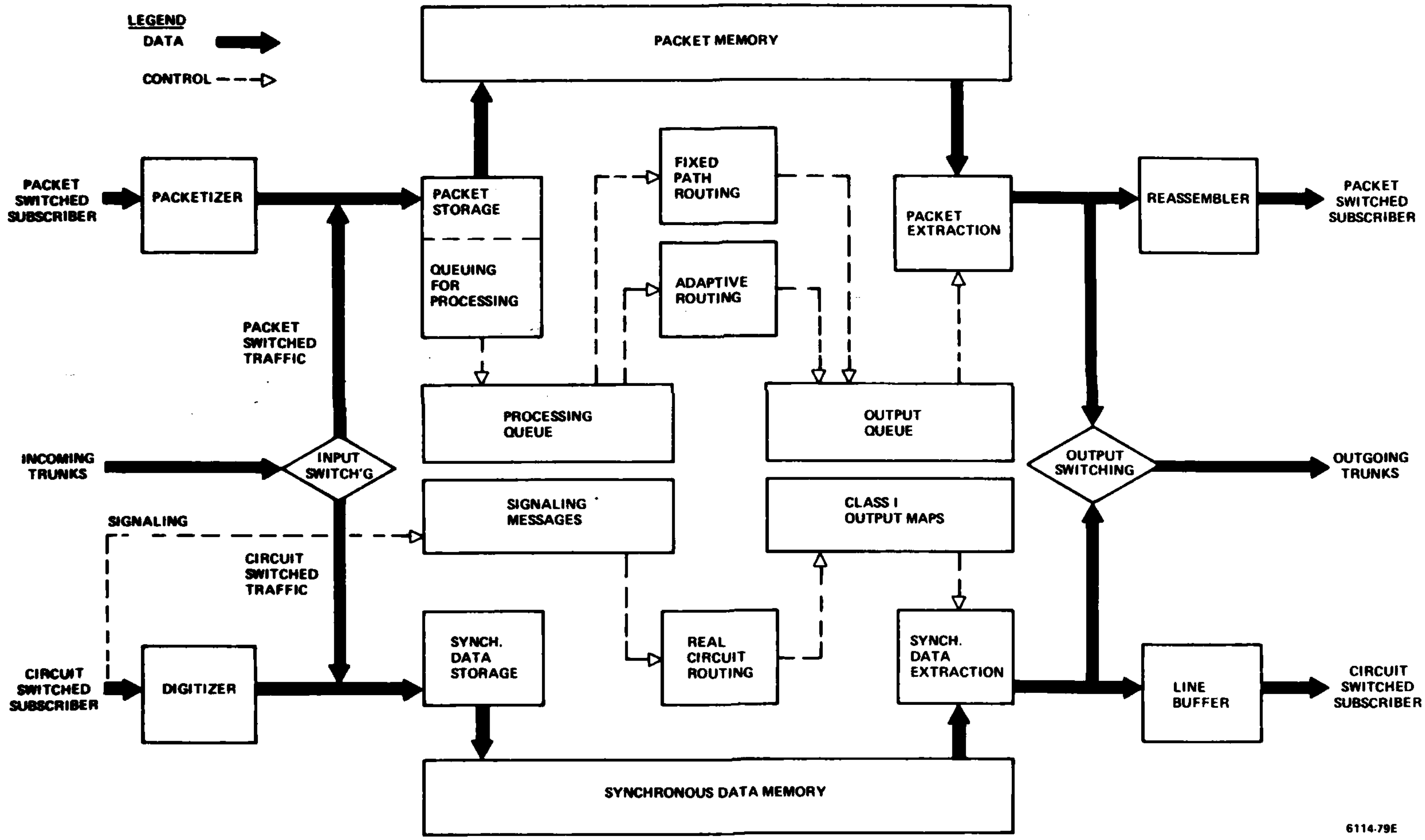


Figure 3.3- Circuit- and Packet-Switched Data Flow at a Flexible Hybrid Switch

delivered asynchronously through the network, but must be delivered synchronously to the destination listener in order to produce intelligible speech. This tends to add to voice delay in the network.

Under the Flexible Hybrid approach, both voice and data can be packetized, sent through the network, and reassembled into complete messages or voice streams at the destination; however, different packet sizes, formats, and protocols may be used for speech and data, depending on the application.[16]

Packet format for data includes flag, address, control, and error-checking fields. The error control requirements for voice packets are less stringent than those for data packets. Since voice can tolerate a higher error rate than data, the check field can be eliminated. If a low percentage of bit errors occurs (on the order of 0.5%), the effect on the transmitted voice quality is usually not objectionable.[4] In any event, voice delay constraints will generally preclude the retransmission of packets. In addition, the sending address can be eliminated, since confirmation of packets is not necessary or desirable. These simplifications will improve the delay performance of packet switching for speech.

Within packet switch processing (see Figure 3.3-6), incoming packets are first stored in memory through a packet storage process. At the same time, a pointer to the stored packet is placed in an input processing queue. The received packets are then subsequently processed in the order of their arrival and routed by the appropriate algorithm based on traffic type. In the Flexible Hybrid concept, voice packets are routed over a virtual circuit by a fixed path routing algorithm, whereas interactive data packets are routed independently by an adaptive routing algorithm. Bulk data is handled by circuit switch processing described below. The routing algorithms place the stored packet pointers in the appropriate output queue; the packets themselves remain in memory where they were stored. Following the examination of pointers in the output queue, packets are extracted from memory and either dispatched over an outgoing trunk to the next switch or delivered to a locally terminated packet switched subscriber.

Within circuit switch processing (see Figure 3.3-6) incoming data is delivered directly to memory via input hardware. Prior to the transfer of actual data, a setup procedure will define the storage locations associated with a particular transaction, and the incoming and outgoing line or trunk. Memory addresses and data lengths are transferred to the output process at the point where the data is to be outputted. Data can then be extracted from its predetermined location in memory and either transmitted over an outgoing trunk or delivered to a locally terminated circuit switched subscriber.

Originating circuit switched traffic that cannot seize channel capacity upon arrival is blocked (managed on a loss basis) while packet switched traffic that is unable to be transmitted is buffered (managed on a delay basis). The boundary between packet or circuit switching may be "fixed" so that no dynamic sharing between traffic classes can take place, or "movable", whereby packet data can seize currently idle circuit switch capacity during a particular frame. Actual interaction between classes of traffic is also a function of the precedence handling associated with each traffic class in priority-based applications. Signaling traffic used in the setting up and breaking down of circuit switched calls is to be packet switched.

3.3.2.3 Optional Switching Models

In addition to the two recommended switching models, (i.e., the MIXD packet switching model and the Flexible Hybrid switching model) there are three additional switching models that deserve further consideration:

a. The all PAR packet switching model
b. The all PVC packet switching model
c. The SENET hybrid switching model

Table 3.3-2 shows the various functions required for implementation of each of the recommended concepts, along with the functions required to implement the optional packet and hybrid concepts. This table shows that by implementing the MIXD and Flexible Hybrid concepts, relatively few new functions are required to implement the PVC,

TABLE 3.3-2. FUNCTIONS REQUIRED FOR IMPLEMENTING PACKET AND HYBRID INTEGRATED VOICE/DATA SWITCHING CONCEPTS

CONCEPTS / FUNCTION	RECOMMENDED CONCEPTS		OPTIONAL CONCEPTS		
	MIXD	FLEX. HYBRID	ALL PAR	ALL PVC	SENET
LINK LEVEL DATA PROTOCOL	●	●	●	●	●
DATA PACKET INPUT HANDLING	●	●	●	●	●
DATA PACKET OUTPUT HANDLING	●	●	●	●	●
ADAPTIVE ROUTING	●	●	●	●	●
ROUTING TABLE UPDATING	●	●	●	●	●
PREC. SCHEMES RECONCILIATION	●	●	●	●	●
TRANSMISSION PRIORITY MAPPING	●	●	●	●	●
PRIORITY QUEUE MANAGEMENT	●	●	●	●	●
SWITCH RESOURCE PREEMPTION	●	●	●	●	●
DATA SILENCE DETECTION	●	●	●	●	●
DATA ACTIVITY DETECTION	●	●	●	●	●
VARIABLE B/W RESERVATION	●	●		●	●
VIRTUAL CONNECTION SU/TD (DATA)	●	●	●		●
VARIABLE PATH DATA PROTOCOL	●	●	●		●
DATA PACKET REASSEMBLY	●	●	●		●
VOICE SILENCE DETECTION	●	●	●	●	
VOICE ACTIVITY DETECTION	●	●	●	●	
LINK LEVEL VOICE PROTOCOL	●	●	●	●	
VOICE PACKET INPUT HANDLING	●	●	●	●	
VOICE PACKET OUTPUT HANDLING	●	●	●	●	
VOICE PACKETIZING	●	●	●	●	
VOICE DEPACKETIZING	●	●	●	●	
VOICE GAP CONTROL (PACKET)	●	●	●	●	
VOICE PACKET DISCARD	●	●	●	●	
VIRTUAL CKT ROUTING	●	●		●	
VIRTUAL CKT B/W PREEMPTION	●	●		●	
FIXED PATH VOICE PROTOCOL	●	●		●	
DUPLEX VIRTUAL CKT SU/TD (VOICE)	●	●		●	
FIXED PATH DATA PROTOCOL	●			●	
SIMPLEX VIRTUAL CKT. SU/TD (DATA)	●			●	
SYNCHRONOUS DATA CCIS PROTOCOL		●			●
SYNCHRONOUS DATA INPUT HANDLING		●			●
SYNCHRONOUS DATA OUTPUT HANDLING		●			●
CCIS INPUT HANDLING		●			●
CCIS OUTPUT HANDLING		●			●
DETERMINISTIC ALT. ROUTING		●			●
VARIABLE B/W ALLOCATION		●			●
REAL CIRCUIT ROUTING		●			●
REAL CKT. B/W PREEMPTION		●			●
START OF FRAME GENERATION		●			●
START OF FRAME DETECTION		●			●
CLASS I/II BOUNDARY DETECTION		●			●
CLASS I MAP MAINTENANCE		●			●
CLASS I COMPACTION		●			●
FRAME PADDING		●			●
VARIABLE LENGTH FRAME MGMT.		●			
SIMPLEX REAL CKT. SU/TD (DATA)		●			
VARIABLE PATH VOICE PROTOCOL			●		
VIRTUAL CONNECTION SU/TD (VOICE)			●		
VOICE REASSEMBLY			●		
DUPLEX VIRTUAL CKT. SU/TD (DATA)				●	
SYNCHRONOUS VOICE CCIS PROTOCOL					●
SYNCHRONOUS VOICE INPUT HANDLING					●
SYNCHRONOUS VOICE OUTPUT HANDLING					●
DUPLEX REAL CKT SU/TD (DATA)					●
DUPLEX REAL CKT SU/TD (VOICE)					●
PARTIAL PACKET HANDLING					●

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PAR, and SENET concepts. The penalty incurred, in terms of the number of functions required for implementation, would be relatively small; therefore, in order to derive the maximum possible capability from the test bed, this functional commonality should be taken into account during the TSN design phase.

The three optional switching models, which are actually minor functional variations of the two recommended switching models, will be discussed in the following sections.

a. PAR Packet Switching Model - In the all PAR packet switching model, voice and data packets are independently routed to the destination. No path is dedicated for the duration of the call. Each packet is transported across the network to its destination independently of other packets for the same destination. Individual packets can be alternately routed as appropriate. Since packets are independently routed, reassembly of voice and data packets is required at the destination node or in the access area. The PAR concept architecture is shown in Figure 3.3-7 which identifies the functions required to handle the various types of traffic.

The PAR concept requires larger packet headers than those in the PVC concept since each packet carries all necessary information to independently route it through the network. Such increase in packet header is expected to result in reduction in the maximum achievable information bit throughput of the network when compared with similar results for the PVC network under identical conditions. One advantage of the PAR concept is its responsiveness to failures in the network by routing packets around the failed paths.

As shown in Table 3.3-2, the all-PAR packet switching model requires three additional functions to be added to the MIXD packet switching model. Note, also, that there are seven functions that are required for the MIXD packet switching model, which are not required for the PAR packet switching model.

The three additional functions required for the all-PAR packet switching model are:

1. Variable path voice protocol, which is responsible for the exchange of voice packets between source and destination nodes of a virtual connection;
2. Virtual connection set-up/teardown for voice calls, responsible for setting up and tearing down the Bookkeeping Blocks (BKBs) associated with the virtual voice connection;
3. Voice packet reassembly, which is responsible for reordering of voice packets at the destination nodes.

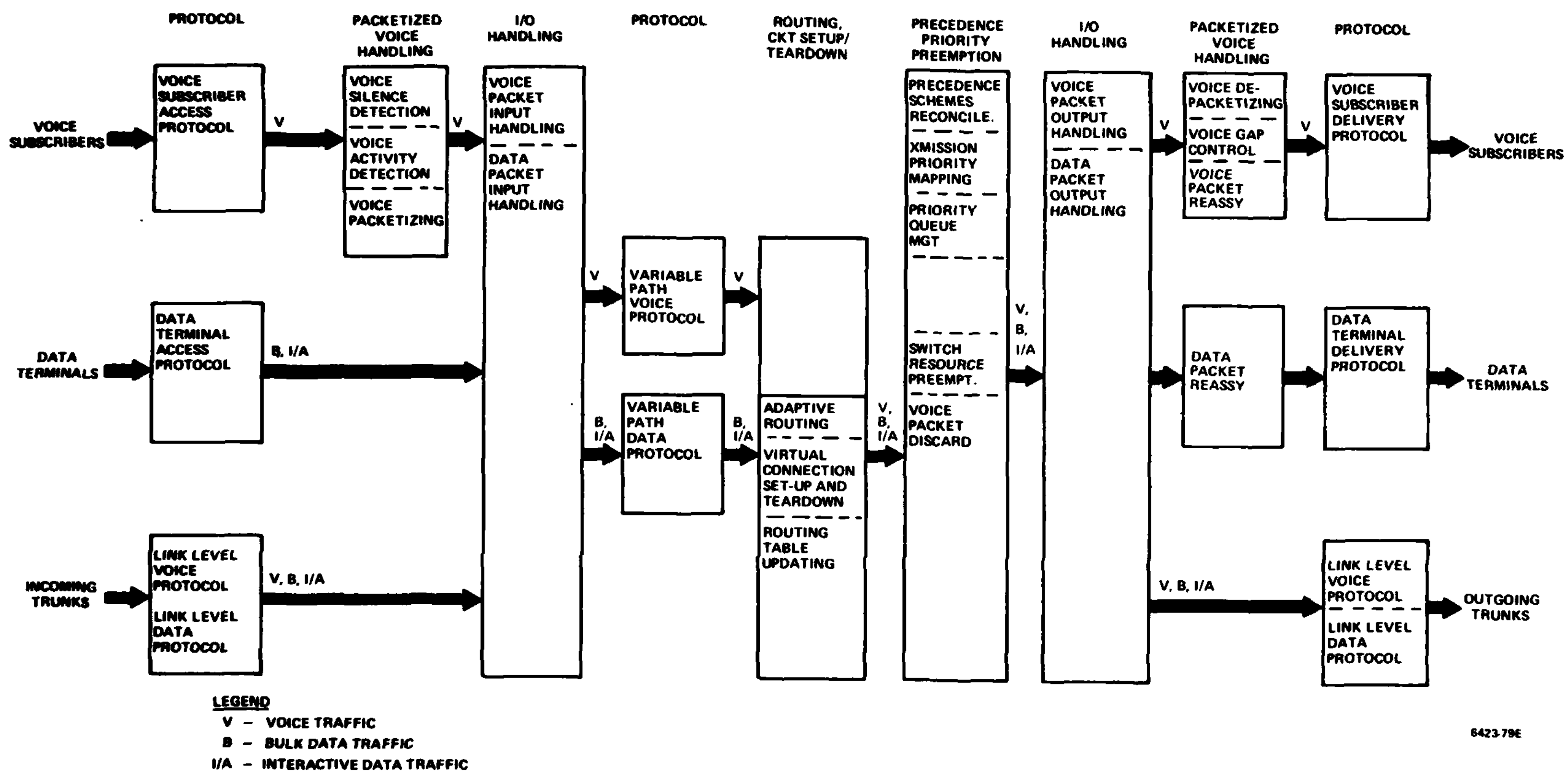

Figure 3.3-7. PAR Concept Architecture

b. PVC Packet Switching Model - In the all-PVC packet switching model, a route is established through the network during a call-initiation period, and the same physical path is used by all packets for the call. At the call-initiation time, a signaling packet, (i.e., "call request" packet) is originated by the source switch and sent to the destination switch where, after verifying that a virtual circuit path can be established for the voice or data call, a call confirmation packet is originated and returned to the source switch. Along the path, data structures are setup at each switching node to specify the outgoing link for every call. All subsequent packets during the call or data transfer transit that fixed path. Upon termination of the connections, the path is released. The PVC concept architecture is shown in Figure 3.3-8, which identifies the functions required to handle the various types of traffic.

The fixed path strategy guarantees that the packets will arrive at the destination in sequence. It also rquires a virtual connection, analogous to that of circuit switching, to be setup between the source and destination and maintained for the duration of the conversation. It can reduce the amount of processing required for individual packets at intermediate switches, since repetitive information such as call identification and routing could be stored at intermediate nodes. A simplified header, indicating the call to which the packet belongs, could be sent, which is sufficient for the receiving node to access the prestored information needed to process that packet. The simplified header can further improve delay performance and shorten the time required to create packets. An added overhead in the fixed-path approach is that the virtual circuit needs to be reestablished every time packets are rerouted, (e.g., in the presence of link and node failure).

As shown in Table 3.3-2, the all-PVC packet switching model requires one additional function to be added to the MIXD packet switching model. Note, also, that there are three functions that are required for the MIXD packet switching model, which are not required for the PVC packet switching model.

The additional function required for the all-PVC packet switching model is the duplex virtual (for data) setup/teardown function. This function is responsible for setting up and tearing down the data structure required for an interactive data call.

c. SENET Hybrid Switching Model - The SENET hybrid switching model is a variation of the Flexible Hybrid switching model; however, instead of packet switching the voice traffic as in the Flexible Hybrid model, the SENET model incorporates the circuit switching of voice traffic. The SENET concept architecture is shown in Figure 3.3-9, which identifies the functions required to handle the various types of traffic.

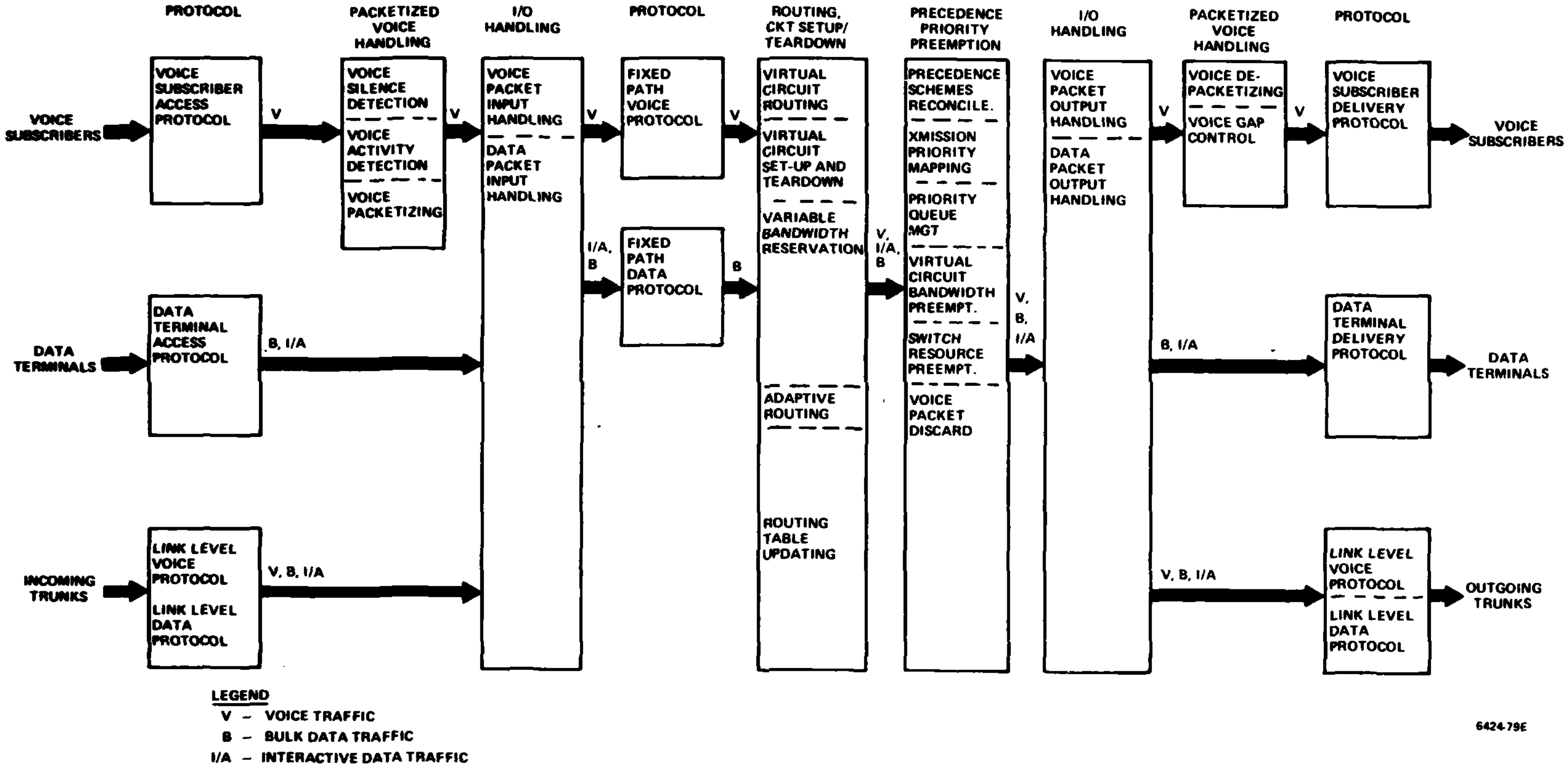

Figure 3.3-8. PVC Concept Architecture

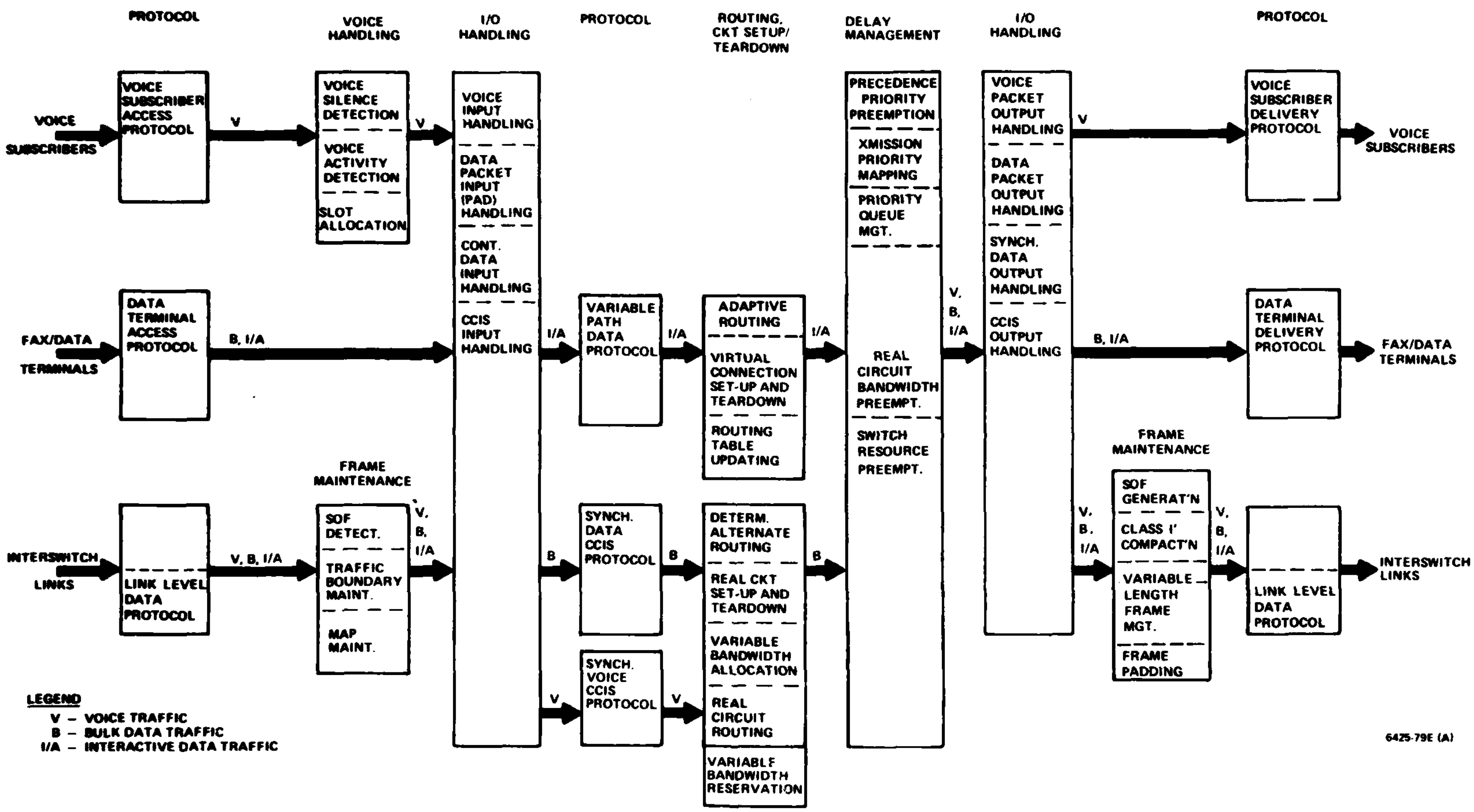

Figure 3.3-9. SENET Concept Architecture

As shown in Table 3.3-2, the SENET hybrid switching model requires six additional functions to be added to the Flexible Hybrid switching model. Note, also, that there are 10 functions that are required for the Flexible Hybrid switching model, which are not required for the SENET hybrid switching model. The six additional functions required for the SENET hybrid switching model are:

1. Synchronous voice CCIS protocol, which is responsible for establishing a real circuit path through the network for circuit switched voice calls;
2. Synchronous voice input handling, which is responsible for steering incoming synchronous voice information to the appropriate buffering area of memory;
3. Synchronous voice output handling, which is responsible for fetching voice information from the appropriate input buffers for transmission on outgoing links;
4. Partial packet handling, which is responsible for handling fragmented packets due to a given packet exceeding the synchronous frame boundary;
5. Duplex real circuit (for data) setup/teardown, which is responsible for setting up and tearing down the data structure for a bidirectional bulk data call using the Class I portion of the SENET frame;
6. Duplex real circuit (for voice) setup/teardown, which is responsible for setting up and tearing down the data structure for a voice call using the Class I portion of SENET frame.

3.3.3 Operational Strategies

This paragraph describes the various operational strategies as they would be recommended for implementation in the MIXD packet switching model and the Flexible Hybrid switching model.

3.3.3.1 Protocol and Procedures

a. MIXD Concept - Protocols describe the rules and procedures that govern the controlled exchange of information between the components of a network. To render different protocol levels independent of each other and lower level protocols transparent to those of the higher levels, a layering approach is adopted as shown in Figure 3.3-10. At the lowest level of present concern is the link protocol layer consisting of the Advanced Data Communication Control Procedure (ADCCP) protocol. This communication discipline constitutes the link protocol between the access devices, such as host, terminal, or vocoder, and the access node as well as between tandem nodes.

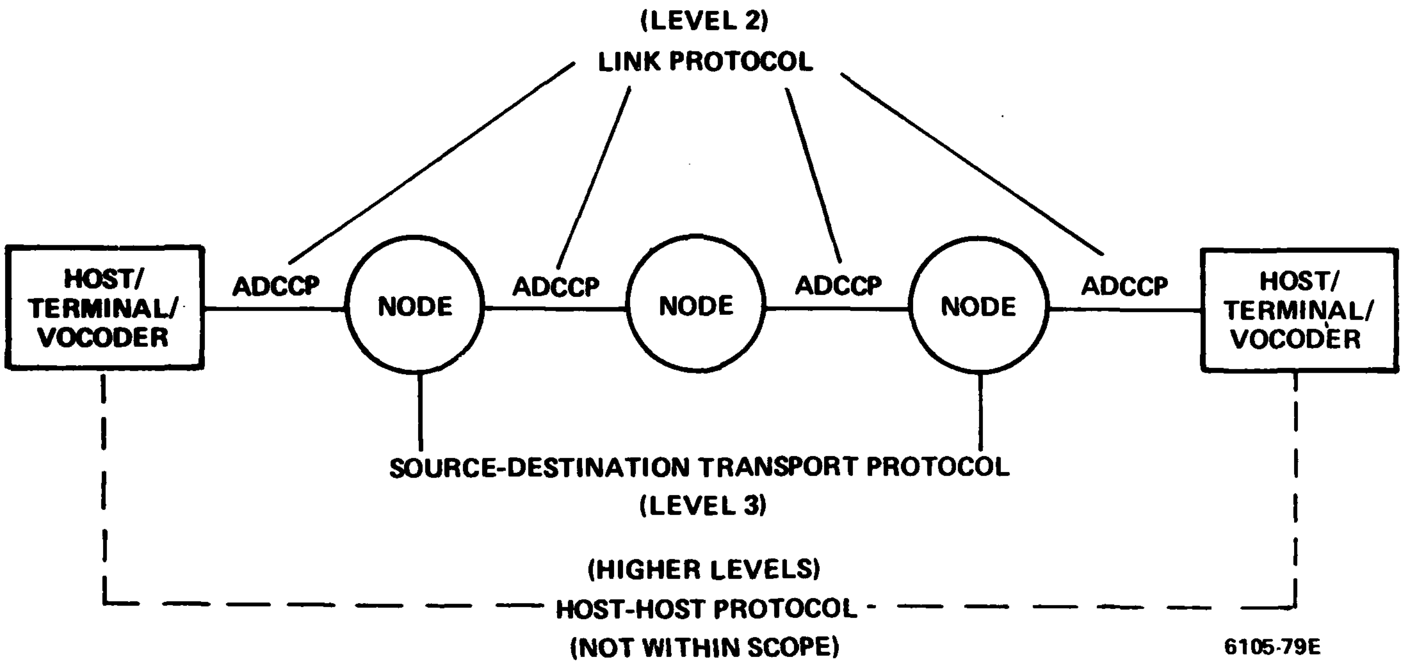

Figure 3.3-10. Protocol Layers for Packet Switching

The source-to-destination node protocol dynamically establishes logical connections between the source and destination nodes. It provides for packet accountability and flow control between the two end-nodes. The tandem node protocol is embedded in the source-to-destination node protocol; both use the packet header for the exchange of protocol information.

The link level and transport protocols are completely transparent to the host-to-host protocol, (i.e., the host and/or terminal communication is not concerned with the mechanics of the lower level protocols). The host-to-host protocol is not within the scope of the ILRAN/TSN program and is only mentioned here to place the protocol layers in perspective.

1. Link Level Protocol - The ADCCP protocol governs the exchange of information at the link level. The protocol execution will distinguish between voice and data type of information. The ADCCP protocol functions will be used extensively for the transmission of data, but only a small subset of these functions will be used for the transmission of voice type of information (speech, facsimile, etc.). The link level protocol for data packets will employ the balanced version of ADCCP described in the ANSI ADCCP standard (ANSI Document X3S34/539);

2. Source Destination Transport Protocol - The source destination protocol depends on the type of information that is to be transported under the MIXD concept. When voice type information (speech, facsimile) is presented to the node, a Fixed Path Protocol (FPP) establishes a path through the network by dispatching a trailblazer packet to the destination. The trailblazer has the effect of creating forwarding tables at each node that it traverses. When the source receives an ACK (trailblazer response) from the destination, it will begin dispatching the packets. No acknowledgments will be returned for voice packets at the transport protocol level;

   The PAR discipline will be employed for routing data packets to the destination. No constant path is established for the duration of the connection. Each packet is transported across the network to its destination independently of other packets for the same connection. In each packet is included a header that contains all the necessary information for traversing the network through the PAR algorithm. The data packet transmission employs the Virtual Flow protocol in which the source maintains in a data structure an account of the data packets transmitted. This data structure is called a Bookkeeping Block (BKB). Since the data packets may traverse the network over different paths and, consequently, may encounter different delays, they

must be resequenced in proper order. This resequencing will be done by the host-to-host level protocol. The Virtual Flow protocol will use acknowledgment by exception, (i.e., only lost or discarded packets will be reported back to the source.

b. Flexible Hybrid Concept - Common Channel Signaling (CCS) has been chosen as the protocol/procedure for handling the assignment and control of the Class I region for the Flexible Hybrid concept.

The particular version of CCS that has been chosen for the ILRAN/TSN network is based on a subset of Federal Standard F6; a federal version of the CCITT Signaling System No. 6. F6 expands the basic CCITT No. 6 to include services and network features unique to federal agencies and departments. The F6 signaling system is basically compatible with the system adopted by the U.S. common carriers, which would not inhibit interoperation in periods of national emergency.

Reliability and flexibility are also important advantages of CCS. The error control provided gives CCS the capability of more reliable transfer of address information than present conventional methods. A particularly outstanding characteristic of CCS, relating to network operation and subscriber service, is that the CCS message format allows considerable latitude and flexibility in transmitting all types of signaling information including signals that might be used for future services not yet defined.

The assignment and control of Class I calls will be accomplished through the use of the CCS messages transported as packets in the Class II region. The standard CCS messages will be embedded in packets that have been formatted using standard ADCCP formats and will be transported using the standard ADCCP link control protocol. Consequently, CCS messages are technically Class II messages; however, to ensure efficient, timely processing of Class I traffic, the CCS messages will be handled at a very high precedence level. The precise level will be determined as part of the TSN experimentation. The CCS portion of the master frame will then tend to precede all the other Class II data packets as shown in Figure 3.3-11.

Note that the Class II packet switched traffic will use the MIXD packet switching protocol/procedures strategy discussed in paragraph a.

3.3.3.2 Precedence and Priority

This section defines the precedence, priority and preemption strategies that are to be incorporated in the various packet switching and hybrid switching concepts to be evaluated. Precedence, priority and preemption are defined as follows: (1) precedence - an inherent

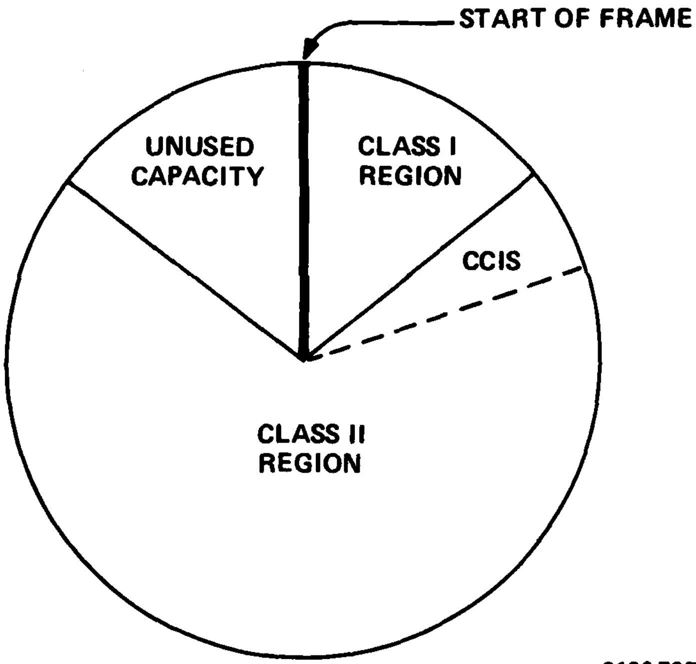

Figure 3.3-11. Impact of CCS on Master Frame Structure

quality of a connection or data, and is assigned by the user; (2) priority - the order in which entities are selected to begin use of network resources; (3) preemption - a process whereby an entity actually using a network resource is dispossessed and the resource given to another entity.

a. General Discussion - All network subscribers will define a precedence level designator associated with their traffic. It is anticipated that most of the data subscribers will also define criticality and application type designators associated with their traffic, whereas voice subscribers will not. The application type designation for voice subscribers will be inherent in their classmarking. The criticality type will be derived from a mapping rule stored in each switch.

1. Precedence - Precedence levels provide an indication of relative importance of traffic in competing for network resources such as link transmission bandwidth and memory space for buffers and queues, and also in cases in which a called subscriber or equipment is busy with another call.

Precedence levels are used in the following ways:

(a) Placing calls in the presence of congestion

(b) Protecting established calls from being preempted by other traffic not of greater importance

(c) Establishing the criticality, (i.e., blocking or nonblocking) types of certain kinds of traffic.

Two separate precedence schemes will be provided in both the MIXD and Flexible Hybrid concepts: a multilevel (five) voice precedence scheme and a multilevel (six) data precedence scheme. These precedence schemes must be reconciled with each other for purposes of blocking and preemption strategy.

2. Criticality and Priority - Originating data subscribers associate with their traffic two important designators. The first is a criticality designator, and the second is an application type designator. Each switch has stored in it a mapping rule which, for every combination of the designators, will assign a priority level. The mapping rule depends on the exact traffic mix of the different traffic types and presents, for the integrated network, a control variable to act on in order to control the relative speed with which a certain traffic type is to be handled. It is desirable to have such a mapping done in a way that both the delay and the continuity requirements for each traffic

type may be satisfied. The criticality and application designators for voice subscribers will be derived as indicated earlier in this paragraph. The system supports the following criticality classes:

Type 1	Nonblocking
Type 2	Blocking

The system will also support the following four application types:

Real-Time - Characterized by long messages requiring low delay, real-time delivery. For example, voice and facsimile subscribers.

Non Real-Time - Characterized by two subclasses.

(a) Low Delay - characterized by two subclasses:

(1) Interactive: characterized by short discrete messages requiring low delay, near real-time delivery wherein the continuity of the information transfer process is maintained during the "session". This category is dominated by interactive data traffic.

(2) Query/Response: the exchange of a question and answer with no attempt to sustain the continuity of the process.

(b) High Delay - characterized by long messages requiring neither continuity nor immediate delivery. This application type consists of the following two subclasses:

(1) Narrative: the transmission of narrative messages in a charcter-oriented format (Average length of 1.2×10^4 bits).

(2) Bulk: Which may vary over the range of the transfer of entire files, programs or processing results having an average transaction length of 5×10^5 bits to the transfer of extremely lengthy information, such as an entire data base with an average transaction length of 3×10^7 bits.

Control Messages - These are the messages exchanged between the different ILRAN nodes to perform the network control procedures.

Burst Mode - Characterized by short bursts (average duration less than 1s) of wide bandwidth real-time information.

3. Preemption - The preemption strategies employed will be a function of the resource that causes the congestion. These strategies differ from each of the switching concepts to be tested. The preemption strategies will consist of various combinations of the following functions: preemption of existing calls, delayed initiation of new calls, discarding of voice packets, data flow control (link-by-link), blocking of new calls, and increased transmission delays.

   The above functions will be used individually or sequentially in situations when the following resources become saturated or otherwise unavailable: pooled resources, (e.g., call stores), data memory or queue memory, transmission bandwidth, called subscriber unavailable (busy), and data throughput (excessive packet delay).

b. MIXD Strategy - The MIXD concept will incorporate the strategies and recommendations discussed in the preceding paragraph. Basically these consist of:

   1. A precedence mapping function that will:

      (a) Reconcile precedence levels among all traffic application types

      (b) Derive a criticality designation for voice subscribers;

   2. A criticality mapping function that will derive a priority level designation for all combinations of criticality types and application types;

   3. A priority queuing function that will select the appropriate outgoing priority queues for voice and data traffic;

   4. A bandwidth preemption function that will preempt bookkeeping blocks (BKBs) for data traffic, or preempt bandwidth reserved for voice traffic.

c. Flexible Hybrid Strategy - a concept that will incorporate strategies and recommendations discussed in paragraph a. Basically these consist of:

   1. A precedence mapping function that will:

      (a) Reconcile precedence levels among all traffic application types

      (b) Derive a criticality designation for voice subscribers;

2. A criticality mapping function that will derive a priority level designation for all combinations of criticality types and application types of packet switched traffic;

3. A priority queuing function that will select the appropriate outgoing priority queues for voice and packet switched data traffic;

4. A bandwidth preemption function that will preempt bookkeeping blocks (BKBs) for PAR traffic, or preempt bandwidth reserved for voice traffic and bandwidth allocated to Class I data traffic.

3.3.3.3 Routing

This section describes the routing strategies that will be incorporated in the MIXD and Flexible Hybrid concepts to be implemented in the TSN.

a. General Discussion - The routing strategy for each concept will consist of a set of basic routing techniques used for routing the various categories (application types) of traffic to be carried. These basic categories are discussed below, in addition to the issues of bandwidth reservation/ allocation and of routing table structure and update.

Three basic routing strategies will be employed in the implementation of the MIXD and Flexible Hybrid concepts:

1. An adaptive packet routing strategy in which packets are each individually routed, independent of other packets. This is the strategy used in the PAR switching concept and will be referred to as the PAR routing technique. Route selection will be based on dynamically updated, minimum delay route tables;

2. A fixed-path packet routing strategy in which a fixed-path virtual circuit is established through the network during a call initiation period, and the same physical path is used by all packets for the call. This is the strategy used in the PVC switching concept and will be referred to as the PVC routing technique. The initial circuit route selection will be based on dynamically updated, minimum delay route tables;

3. A fixed-path synchronous traffic routing strategy in which a fixed-path synchronous (or "real") circuit is established through the network during a call initiation period and remain dedicated for the duration of the call. The route selection will be based on static deterministic tables containing primary and alternate routes; hence the name, deterministic alternate routing (DAR) technique.

These three basic strategies may in fact be used in various combinations for any of the switching concepts studied during the Concept Evaluation phase.

In the test bed environment, a fourth strategy is required to handle certain types of self-routing packets used for delay measurements or continuity testing of prespecified paths. The technique that will be used will entail embedding the routing information within the packets (PER). An alternate technique would be to artificially bias the transmission queues at the appropriate nodes in such a way that the packets would be adaptively routed along the desired path. This technique would have the virtue of not requiring an additional routing algorithm at the nodes, but it would tend to distort the network flow patterns and therefore will not be used.

A distinction is made between bandwidth reservation and bandwidth allocation. Bandwidth that is reserved for a call may be utilized temporarily for transmitting packets associated with other calls or control packets, but may not be reserved for or allocated to another call. Allocated bandwidth is set aside for the exclusive use of the call to which it is allocated.

Reserved bandwidth and allocated bandwidth both are used in determining if a link has sufficient capacity to handle an additional Class I call or PVC call. The MIXD concept uses bandwidth reservation whereas the Flexible Hybrid concept uses both bandwidth reservation and bandwidth allocation.

b. <u>MIXD Concept Routing Strategy</u> - Under the MIXD concept, traffic requiring real-time or near-real-time delivery will be handled by the PVC routing technique in order to obtain the best continuity and delay performance with minimum overhead. Nonreal-time traffic will be handled in either of two ways, depending on message length. Such traffic involving short messages will be handled by the PAR routing technique, which tends to reduce local congestion buildup in the network. Nonreal-time traffic involving long messages will be handled by the PVC method to reduce transmission overhead. Therefore the identification of traffic application type will be important in determining which routing procedure an arriving packet will follow. Table 3.3-3 defines the routing techniques that will be used for the various traffic application types in the MIXD concept.

Table 3.3-4 defines the bandwidth reservation strategies that will be used with the various traffic application types in the MIXD concept. These are described below:

1. Voice, facsimile, and bulk traffic, which are PVC routed, have bandwidth reserved based on the average baud rate. This is accomplished when the trailblazer for the call is setup. Bandwidth is not allocated, but is reserved for PVC calls;

TABLE 3.3-3. MIXD ROUTING TECHNIQUES

Class	Application Type		Routing Technique
	Delay	Subclass	
Real Time	Very Low	Voice	PVC
		Facsimile	
Non-Real Time	Low	Interactive	PAR
		Query/Response	
	High	Narrative	
		Bulk 1	PVC
		Bulk 2	
Burst	Low	Wideband Real Time Information Bursts	
Control	Low	Trailblazers, Flow Control, etc.	PVC, PAR, PER

PVC - Packet Virtual Circuit
PAR - Packet Adaptive Routing
PER - Packet Embedded Routing

TABLE 3.3-4. MIXD CONCEPT BANDWIDTH RESERVATION

APPLICATION TYPE			BANDWIDTH RESERVED		BANDWIDTH ALLOCATED	
CLASS	DELAY	SUBCLASS	DIRECTION	WHEN	DIRECTION	WHEN
REAL TIME	VERY LOW	VOICE	DUPLEX	WHILE ROUTING TRAILBLAZER	BANDWIDTH NEVER ALLOCATED	
		FACSIMILE	SIMPLEX			
NON-REAL TIME	LOW	INTERACTIVE	BANDWIDTH NOT RESERVED			
		QUERY/RESPONSE				
	HIGH	NARRATIVE				
		BULK 1	SIMPLEX IN DIRECTION OF TRANSFER	WHILE ROUTING TRAILBLAZER		
		BULK 2				
BURST	LOW	WIDEBAND REAL TIME INFO. BURSTS				
CONTROL	LOW	TRAILBLAZERS, FLOW CONTROL ETC.	BANDWIDTH NOT RESERVED			

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2. Interactive, query/response, and narrative traffic is handled by PAR techniques, and requires neither reservation nor allocation of bandwidth;

3. Control traffic requires neither reservation nor allocation of bandwidth.

c. Flexible Hybrid Routing Strategy - Under the Flexible Hybrid approach, traffic requiring real-time (or near-real-time) delivery will be handled in either of two ways. Real-time traffic that is essentially undirectional and has a low peak-to-average bandwidth, (i.e., not bursty) will be handled as Class I traffic and routed over simplex real circuits via the deterministic alternate routing technique. Bidirectional and bursty real-time traffic will be handled as Class II traffic and routed via the packetized virtual circuit technique.

This strategy is expected to improve delay performance of the undirectional nonbursty traffic and decrease processing bandwidth without the penalty in transmission efficiency that would be associated with duplex real circuits.

Nonreal-time traffic will also be handled in either of two ways, depending on message lengths. Lengthy data calls, which are essentially undirectional and which have a low peak-to-average bandwidth ratio (i.e., not bursty), will be handled as Class I calls and routed over simplex real circuits via the deterministic alternate routing technique. This strategy is expected to decrease processing bandwidth and increase transmission efficiency because of the reduction of packet overhead.

Table 3.3-5 defines the basic routing techniques that will be used for the various traffic application types in the Flexible Hybrid concept.

Table 3.3-6 defines the bandwidth reservation and allocation strategies that will be used with the various traffic application types in the Flexible Hybrid concept. These are described below:

1. Voice traffic, which is PVC routed with a full duplex virtual cricuit, has bandwidth reserved based on the average baud rate of the call. This is accomplished when the data structure for the call is setup while routing the trailblazer. Bandwidth is not allocated for PVC calls;

2. Facsimile calls have bandwidth reserved along the simplex real circuit while the call is being setup in the forward direction. This reserved bandwidth is subsequently allocated when the called party goes off-hook. This technique is referred to as "forward reservation, delayed backward allocation". The

TABLE 3.3-5. FLEXIBLE HYBRID ROUTING TECHNIQUES

Application Type			Routing Technique
Class	Delay	Subclass	
Real Time	Very Low	Voice	PVC
		Facsimile	DAR
Non-Real Time	Low	Interactive	PAR
		Query/Response	
	High	Narrative	DAR
		Bulk 1	
		Bulk 2	
Burst	Low	Wideband Real Time Information Bursts	
Control	Low	Trailblazers, Flow Control, etc.	PVC, PAR, DAR, PER

PVC - Packet Virtual Circuit
PAR - Packet Adaptive Routing
DAR - Deterministic Alternate Routing
PER - Packet Embedded Routing

TABLE 3.3-6. FLEXIBLE HYBRID BANDWIDTH RESERVATION/ALLOCATION

Application Type			Bandwidth Reserved		Bandwidth Allocated	
Class	Delay	Subclass	Direction	When	Direction	When
Real Time	Very Low	Voice	Duplex	While Routing Trailblazer	Not Allocated	
		Facsimile	Simplex		Simplex in Dir. of Transfer	While Routing Answer MSG Back Over Circuit
Non-Real Time	Low	Interactive	Bandwidth Neither Reserved Nor Allocated			
		Query/Response				
	High	Narrative				
		Bulk 1	Simplex in Direction of Transfer	While Routing Call Setup CCIS MSG	Simplex in Direction of Transfer	While Routing Ans. MSG Along Virtual Circuit
		Bulk 2				
Burst	Low	Wideband Real Time Info. Bursts				
Control	Low	Trailblazers, Flow Control, etc.	Bandwidth Neither Reserved Nor Allocated			

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acknowledgments, which are packet switched, require neither reservation nor allocation in the backward direction;

3. Interactive, query/response, and narrative traffic is handled by PAR techniques and therefore requires neither reservation nor allocation of bandwidth;

4. Bulk 1 and Bulk 2 calls are treated like facsimmile calls;

5. Burst traffic is handled like facsimile traffic, except that the Class I/Class II boundary will be allowed to move beyond the normal limit due to the short duration of the wideband bursts.

6. Control traffic requires neither reservation nor allocation of bandwidth.

3.3.3.4 Network Flow Control

a. General Discussion - The purpose of flow control is to regulate the access and network traffic such that demands on the network are reconciled with available resources. The temporary disruption of a critical component, differences in access circuit speeds at the input and output side of a connection, or a temporary, unforeseen, heavy traffic load converging on a node may cause mismatches between demands and resources. A key assumption in the following discussion is that voice and data type information are maintained in separate queues. This enables each to be managed according to different flow control disciplines.

b. MIXD Concept - Under the MIXD model the voice type information will be subjected to flow controls consistent with the PVC concept, and the data type information will be subjected to flow controls consistent with the PAR concept.

1. Flow Control for Voice Type Connections - For voice circuits the need for flow control will arise due to shortage of buffer space or unavailability of transmission bandwidths, but never due to mismatch of input and output access circuits. There are three ways of exercising limited flow control on a voice type connection:

(a) Variable Rate Encoding - Experimental work is being performed on variable rate encoders, where the voice frames have the property that bits can be systematically "stripped" from the frame without destroying the frame's usefulness. This technique provides the means to restrict the flow of data by dropping bits in a systematic fashion with the least adverse effect on the voice quality. It will be useful to experiment with this technique

in order to establish its practicality for the TSN. The variable rate of encoding presumably could be controlled by the software of the node. Whenever congestion occurs in the voice type information queues, a signal would be fed back to the source vocoder, which would reduce the rate of voice code generated and introduced to the node on this circuit. If the variable rate encoder whould prove practical, it could provide an efficient method of implementing flow control for voice type circuits;

(b) Discarding of Voice Packets - Discarding of voice packets as a method of flow control must be subject to further qualifications with respect to criticality. Blocking type circuits will be subjected to packet discarding when buffer congestion occurs. If the proportion of discarded packets exceeds a certain threshold (about 1% presumably) the quality of the connection may deteriorate significantly. It will then be at the discretion of the subscribers to discontinue the connection when the voice fidelity becomes too poor for maintaining the conversation. Nonblocking circuits may not be subjected to packet discarding that exceeds the permissible acceptability (about 1%);

c. Preemption of Packetized Virtual Circuits - A virtual circuit and its corresponding bandwidth are maintained for the duration of the circuit connection. Consequently, the temporary unavailability of circuit bandwidth cannot be alleviated by discarding packets. Therefore, when a subscriber of high precedence and criticality wants to access the network and bandwidth is unavailable, a blocking virtual circuit with the same address parameters, but lower precedence will be preempted. Choosing a virtual circuit with the same source and destination addresses as the preempting circuit will ensure that not more than one virtual circuit has to be preempted, assuming that the preempted and preempting circuits have the same bandwidth.

2. Flow Control for Data Type Connections - The data access flow to a node is controlled by a window mechanism that allocates to a host or terminal the rate at which packets may be introduced to the network for transmission. Normally, the receiving window is unlikely to be exhausted before the node provides an update. The packets are accepted normally by the node, processed, and queued for transmission on the proper output trunk or line.

The actions taken to exercise flow control on data transport connections can be summarized in the following steps:

At the destination and tandem nodes:

(a) Return Congestion Control Notice to the source node when output or trunk queue exceeds the threshold value (e.g., 75% filled)

(b) Discard blocking packets first when output or trunk queue exceeds the saturation value (100%). Return Non-Delivery Notice to source host.

At the source node:

(a) Respond to CCN messages from destination or tandem nodes by suspending flow of data packets subject to time-out setting

(b) Maintain suspended packets in temporary buffer

(c) Discard blocking undeliverable packets

(d) Send NDN messages to host

(e) Reduce receive window

(f) Resume packet flow on timer expiration.

3. Relational Issues of Voice and Data Flow Control - There will be a maximum reserved bandwidth for voice type connections; there will be no maximum bandwidth for data type connections. When voice type connections reach the maximum reserved bandwidth, there will be no effect upon data type connections. One type of connection can affect the other if there is a precedence imbalance. If all of the connections of one type are of a higher precedence than the other and queue buildup occurs in the higher precedence connection type, it may be necessary to apply flow control to the lower precedence connections. In all cases where flow control must be applied, the decision must be based on the precedence, priority, and preemption scheme and the availability of alternate routing. The recovery from such queue buildup occurrences requires the application of flow control, alternate routing, and preemption.

c. <u>Flexible Hybrid Concept</u> - The Flexible Hybrid model will handle Fax, Bulk and Burst data in the Class I region. Voice, as well as interactive, narrative, etc., will be handled in the Class II region. Voice type connections will be subject to flow controls consistent with the PVC concept. All data type connections in the Class II region will be subject to flow control consistent with the PAR concept. Flow control in the Class I region will be handled similarly to traditional circuit switching on a blocking basis.

1. Flow Control for Voice Type Connections - All voice type connections will be handled in the Class II packet region. Flow control management will be identical to that described for voice type connections under the MIXD model;

2. Flow Control for Data Connections - To optimize utilization of network resources, simplex data such as Fax, Bulk and Burst will be transmitted in the Class I region. These connections have no end-to-end flow control once the link has been established. Management will be during the establishment of a connection. Blocking will occur if resources are not available on the primary or any specified alternate routes. Link-to-link control will be via CCIS messages. Delay management will not be used in the Class I region.

   All other data (Interactive, Narrative, etc.,) will be processed in the Class II region. Flow control management will be identical to that described for data type connections in the MIXD model;

3. Relational Issues between the Class I and Class II Regions - If the frame is completely filled and another connection must be added to one region, it may be necessary to apply flow control to the other region. When this situation occurs, connections can be preempted or alternate routing can be attempted. It is also possible to apply flow control to the other region and wait for bandwidth to become available. Relational issues between voice and data in the Class II region are identical to the relational issues of voice and data control in the MIXD model.

3.3.3.5 Network Link Error Control

a. General Discussion - Error Control is defined as the scheme employed to guarantee a given user-to-user undetected bit error rate through the network for subscribers having error controlled access circuits. Link error control will be provided for the Class II (packet switched) traffic on each leg of the physical path through the network. The link error control provided by the network supplements the end-to-end error control provided by the users. In addition, CCIS traffic has its own end-to-end error control procedures. Error control for both the MIXD packet switching and the Flexible Hybrid switching concepts is based on error detection and retransmission requests, so-called ARQ procedures.

b. MIXD Concept - The MIXD switching concept will employ the error detection and recovery procedures of the ANSI ADCCP protocol for link error control in the transmission of data packets via the PAR discipline. Link error control is not

needed in the transmission of voice packets; consequently, error detection and recovery procedures will be bypassed for voice transmission.

In the transmission of data packets the link protocol employs the balanced class or primary/primary version of ADCCP in which each node is responsible for the correction of errors associated with frames that it transmits. All stations use the Automatic Response Mode (ARM) wherein each station maintains a Send Sequence Number N(S) and Receive Sequence Number N(R). The N(R) value indicates to the local station that the remote station has correctly received all information frames up to and including N(R)-1.

Error detection is accomplished by the ADCCP standard Cyclic Redundancy Check (CRC), which generates a 16-bit Frame Check Sequence (FCS) as the remainder of the division by a 16th degree polynomial. The FCS is transmitted with every information frame to the receiving node. The FCS generated at the receiving node must be equal to the FCS attached to the information frame to ensure that the transmission is error-free. All frames for which the CRCs result in FCS errors are discarded by the receiving node and will be retransmitted by the source node.

The ADCCP document recommends the polynomial $X^{16} + X^{12} + X^5 + 1$ as a divisor and Modulo-2 remainder as the check. This will also be adopted as the standard for the ILRAN/TSN network.

Information frames having sequence number errors, i.e., N(S) that are not equal to the expected sequence number of the receiving station (the R variable), will be discarded. The N(R) value is assumed to be valid.

A Frame Reject Response (FRMR) is initiated upon receipts of a frame containing an invalid command or response, an I-field, which exceeds the maximum length, or an invalid frame format.

An invalid receive sequence number N(R) implies that synchronization has been lost; i.e., the send sequence number variable R at the local station does not match the receive sequence number variable S at the remote station. The station receiving the invalid N(R) will send a RSET command and assign new sequence numbers starting with zero after the RSET command is acknowledged.

Response timers are maintained by each transmitting station to detect, identify and aid in the recovery time-out conditions. The information frame timer solicits acknowledgments of unacknowledged I-frames upon expiration. An unnumbered command timer monitors the responses to unnumbered commands. A supervisory command timer monitors the responses to a supervisory command frame. When these timers expire before receiving a response, the command frames are repeated a

certain number of times (e.g., three times). When this number is exceeded, a message for operator intervention may be displayed or an alarm sounded.

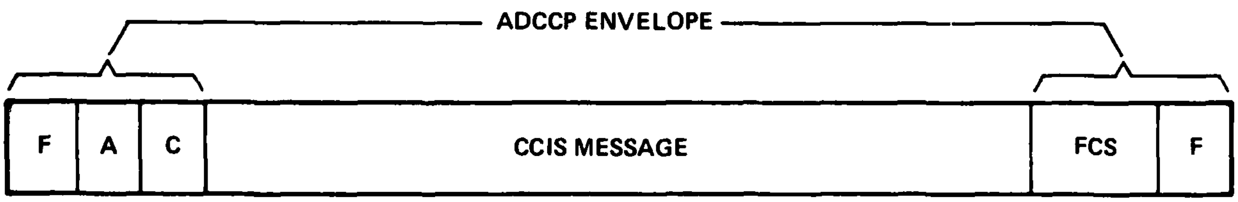
c. Flexible Hybrid Concept - The Flexible Hybrid switching concept separates traffic into two classes: Class I and Class II. Class II traffic (packet switched) error control was described in the preceeding paragraph. The error control associated with the assignment and control of Class I traffic will be described here.

Error control is provided for the CCIS messages that control the assignment of Class I traffic slots. This includes link-by-link as well as end-to-end error control procedures. Error control is also provided for the Class I traffic itself. This is an end-to-end error control procedure between source and destination subscribers with the acknowledgment (ACK) nonacknowledgment (NACK) messages returned to the Class I traffic orginator as Class II (packet switched) traffic.

1. CCIS Error Control - The error control provided for the CCIS messages consists of both link-by-link and end-to-end error control procedures. The link-by-link error control procedure will use the ADCCP protocol, i.e., the CCIS message will be embedded in the ADCCP envelope as shown in Figure 3.3-12a. In addition, an end-to-end error control will be implemented by using the check bits on the CCIS message (see Figure 3.3-12b) to validate the integrity of the CCIS message. Retransmission requests will follow the CCIS acknowledgment strategy described in 3.3.3.6.

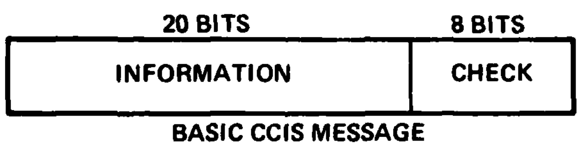
2. Class I Traffic Error Control - The Class I traffic (Bulk 1 data, Bulk 2 data, Burst data, and Fax) may require an end-to-end error control procedure. Some of the Class I subscribers will have error control equipment associated with their terminals. Various forms of user end-to-end error control have been implemented, including Automatic Repeat Request (ARQ) and Forward Error Correction (FEC) techniques. The FEC techniques will have no impact on the ILRAN/TSN architecture or design, since they are user-to-user only and do not require any action by the ILRAN/TSN network.

   The end-to-end ARQ techniques will have an impact on the ILRAN/TSN network. Normal call setup for a Class I call consists of setting up an end-to-end simplex connection for the passage of Class I traffic. The receiving terminal would then perform error detection checks (e.g., horizontal/vertical parity checks) on the Class I traffic. Confirmation of correct reception will be accomplished with an acknowledgment (ACK) message. Incorrect reception will be conveyed by a nonacknowledgment (NACK) message. The ACK/NACK



F = FLAG SEQUENCE
A = ADDRESS FIELD
C = CONTROL FIELD
FCS = FRAME CHECK SEQUENCE

a.) CCIS LINK ERROR CONTROL



b.) CCIS END-TO-END ERROR CONTROL

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Figure 3.3-12. CCIS Error Control Message Structure

messages will be returned from the destination ILRAN to the originating ILRAN as Class II (packet switched traffic.)

3.3.3.6 Acknowledgment

a. General Discussion - Acknowledgment is defined as the ability of the network to furnish information about the delivery status and data integrity of the traffic carried by the network. The acknowledgment strategy for both the MIXD packet switching and the Flexible Hybrid switching concepts is based on error detection and retransmission requests. Information concerning the integrity of the transported messages is conveyed via acknowledgment (ACK) and nonacknowledgment (NACK) messages.

b. MIXD Concept Acknowledgment Strategy - The acknowledgment strategy is based on the assumption that the network only rarely will lose a packet (less than one packet in a thousand should have to be retransmitted). Furthermore, it is assumed that the host-to-host protocol will acknowledge all data type information exchanged between hosts and/or terminals. The voice type information transfer is of the nature of a real-time transmission for which acknowledgment would be redundant and useless, since the voice type packets cannot be repeated.

Consistent with the protocol levels depicted in Figure 3.3-10, the ILRAN/TSN acknowledgment strategy extends over two protocol levels: link protocol (level 2) and network transport protocol (level 3).

1. Link Level Acknowledgment - At the link level, the ADCCP protocol acknowledges the information frames by specifying the latest frame count, N(R), that has been satisfactorily received. N(R) acknowledges all frames having sequence numbers up to and including N(R)-1. ADCCP connections using the basic control field format can send a maximum of seven information frames before an acknowledgment on at least some of the outstanding frames must be received. The line termination handler that controls the link protocol will be designed to differentiate between voice and data packets. Thus, voice type frames will be recognized and no information frame acknowledgments for voice packets will be returned to the transmitter.

2. Network Level Acknowledgment - The main purpose of the network level acknowledgment strategy is to maintain positive control on the arrival of the packets at the destination and to initiate retransmission if the source does not receive an acknowledgment within a certain time-out period. However, network level

acknowledgments of all packets may be highly redundant if the hosts and/or terminals acknowledge the data exchange at the host to host protocol level. Since acknowledgment packets may constitute a significant portion of the network traffic, a redundant acknowledgment strategy that confirms the reception of all arrived packets will be very costly in terms of node and network capacity. Therefore, the method of negative acknowledgment that reports only lost or discarded packets to the originating source is the preferred strategy. This is part of the Virtual Flow protocol (ref. paragraph 3.3.3.1).

c. Flexible Hybrid Acknowledgment Strategy - The Flexible Hybrid switching concept separates traffic into two classes: Class I and Class II. The Class II (MIXD packet switched) acknowledgment strategy is described in b. above. The acknowledgment strategy associated with the assignment and control of Class I traffic will be described here.

The acknowledgment strategy is part of the overall error control strategy for the network. Thus, acknowledgment control is provided for the CCIS messages, which control the assignment of Class I traffic slots. This includes link-by-link acknowledgment as well as end-to-end acknowledgment procedures. Acknowledgment control is also provided for the Class I traffic itself. This is an end-to-end acknowledgment procedure between source and destination subscribers with the acknowledgment (ACK)/nonacknowledgment (NACK) messages returned to the Class I traffic originator as Class II (packet switched) traffic.

1. CCIS Acknowledgment Strategy - The acknowledgment control provided for the CCIS messages consists of both link-by-link and end-to-end acknowledgment procedures. The link-by-link acknowledgment strategy will employ the ADCCP procedures, as discussed in b. above. In addition, an end-to-end acknowledgment associated with the CCIS protocol will be provided.

   The basic CCIS word, or signal unit, which is used to form CCIS messages, consists of 26 bits. Twenty bits contain signaling information and eight bits make up a cyclic check code that will be used for end-to-end error detection. The signal units are transmitted in blocks of 12, the 12th position of every block contains an acknowledgment control unit (ACU). This ACU is coded to indicate the number of the block being transmitted, the number of the block being acknowledged, and whether or not each of the 11 other signal units in the block being acknowledged were received without detected errors. Blocks that contain errors in any of their signal units will then be retransmitted.

2. Class I Traffic Acknowledgment Strategy - The Class I traffic (Bulk 1 data, Bulk 2 data, Burst data, and Fax) may require end-to-end acknowledgment control. Normal call setup for a Class I call consists of setting up an end-to-end simplex connection for the passage of Class I traffic. The receiving/destination terminal would then perform error detection checks (e.g., horizontal/vertical parity checks) on the Class I traffic. Confirmation of correct reception will be accomplished with an acknowledgment (ACK) message. Incorrect reception will be conveyed by a nonacknowledgment (NACK) message. The ACK/NACK messages will be returned from the destination ILRAN to the originating ILRAN as Class II (packet switched) traffic.

3.3.3.7 Delay, Voice Continuity, Error and Loss Strategy

a. General Description - A delay strategy based on the use of precedent and priority will be used to ensure that the delivery requirements for the various traffic classes are met. The priority of transmission, which is the order in which the trunk handler selects queued packets or synchronous data for transmission over a trunk, is based on the priority designation for each type of traffic. Control messages, (i.e., trailblazer packets, CCIS messages, etc.), are given the highest priority. Synchronous traffic, requiring low delay (less than 200 ms and low delay interactive (asynchronous) data traffic will be given the next highest priorities. Other traffic, (i.e., bulk, narrative/record, bursts, etc.), will be given a priority commensurate with meeting the delay requirements listed in Table 3.3-7.

b. MIXD Concept

1. Flow Control - Data flow control, in which a fixed limit is imposed on the size of the data queue waiting to be serviced, will be used to minimize mean delay. This strategy is utilized in addition to the primary strategy of managing delay by precedence and preemption. In the scheme, packets are denied entry to the multiplexed link when a queue threshold is exceeded. The rejected packets are either discarded or retransmitted. In general, voice packets are discarded before data packets, subject to precedence ranking. The node may discard a certain percentage of voice pockets of low precedence calls without seriously impairing the voice intelligibility or quality (0.5%). Beyond this point, it would be up to the subscriber whether or not to continue the conversation at the degraded level. Discarded data packets would be retransmitted when no acknowledgment is received, resulting in additional delay. Thus, the mean delay of packets accepted for

TABLE 3.3-7. END-TO-END DELIVERY DELAY REQUIREMENTS

Type of Transaction	Subcategory	Delivery Time (Min.)	Delivery Time* (Max.)	Average Length
Interactive	Human Interaction	< 1 s	2 s	600/6000 bits
	Alarm/Status Indicators	< 1 s	3 s	2
	Monitoring/ Telemetry	0.3 s	1 s	600
Query/ Response	-	36 s	2 min	600/6000
Narrative/ Record	-	1.6 min	6.4 min	1.2×10^4
Bulk 1	-	1.6 min	6.4 min	5×10^5
Bulk 2	-	4 hr	12 hr	3×10^7

* Corresponds to the 38 point.

transmission is decreased at the expense of those packets that are rejected when the queue threshold is exceeded.

2. Packet Switched Voice Overhead - Under the packet switching concept, both voice and data are packetized, sent through the network, and reassembled into complete messages or voice streams at the receiver. However, different packet sizes, formats, and protocols are used for speech and data, depending on the application. The datagram model for packet switching assumes sufficient control information in the header to allow error detection and retransmission, control, and acknowledgments of received packets. Packet format includes flag, address, control and Cyclic Redundancy Check (CRC) fields. The requirements for voice packets, however, appear to be less stringent than those for data packets. Since voice can generally tolerate a higher error rate than data, the check field can be eliminated. If a low percentage of bit errors occurs (on the order of 1%), the effect on the transmitted voice quality is usually not objectionable. Voice delay constraints will generally preclude the retransmission of packets. In addition, the sending address can be eliminated, since confirmation of packets is not necessary or desirable. These simplifications will improve the delay performance of packet switching for speech.

   Under the fixed path protocol (equivalent to the Packet Virtual Circuit (PVC) approach), a route is established through the network during a call-initiation period. signaling messages, (i.e., "trailblazer" packets), are propragated to the destination, and after verifying that a virtual circuit path can be established for the voice or data call, a call confirmation packet (ACK) is returned. Along the path, pointers setup at tandem switching nodes specify the outgoing link for every packet entering the node. All subsequent packets during the call or data transfer transit that fixed path. Upon termination of the connections, the path is released. During the call or data transfer, no channel capacity is reserved nor is any switch capacity dedicated to the call.

   The fixed path strategy requires a virtual circuit, analogous to that of circuit switching, to be setup between the source and destination and maintained for the duration of the call. Consequently, it guarantees that the packets will arrive at the destination in sequence. This is a significant advantage for maintaining voice continuity. It also reduces the amount of processing required for individual packets at intermediate switches. Repetitive information, such as call identification and routing, is stored at intermediate nodes. A simplified header indicating the call to

which the packet belongs is sent, which is sufficient for the receiving node to access whatever information is needed to process that packet. The simplified header further improves delay performance and shortens the time required to create packets.

Because of the abbreviated header, the fixed path approach is less robust in the presence of link and node failures than the "pure" packet switching approach. In such cases, recovery procedures must be initiated.

An extension of the fixed path strategy is to group together all the packets from the established voice circuits of a particular link into a "super" packet that is transmitted using a single header.[16] Within the header, each voice circuit is identified in a circuit identifier field. If a circuit is active during transmission of the "super" packet, this field has a "1" in the corresponding bit position; if it is inactive, it has a "0". The length of each super packet will be a function of the length of the queue waiting to be transmitted and will vary from packet to packet. This modification retains most of the flexibility and effectiveness of packet switching, but can potentially reduce the overhead close to that of circuit switching. It also adds to delay and delay dispersion, because of the time required to gather the "super-packet" together at one end of the link and separate out the individual packets at the other end.

The added processing complexity and real-time synchronization problems make this approach similar to tandem TASI in the circuit switching concept. Since this added operational complexity would further affect voice continuity, and because the design implementation would be difficult, the technique is recommended for future consideration only.

In the path-independent approach used for data, packets are independently routed to the destination. No path is setup for the duration of the transaction. Each packet is transported across the network to its destination, independently of other packets for the same connection. Individual packets can be alternately routed as appropriate. This "pure" packet approach could result in packets belonging to the same message being received out of order because of the varying delays encountered over different network paths. Hence, a reassembly mechanism is required at the destination. Likewise, the algorithm for adaptive routing of packets is more complex. However, communication is more robust in the event of failure, since packets can be routed around failed paths. This approach makes fewer assumptions about network services than the fixed path strategy, and therefore allows for internetting with a

much wider class of different packet switching networks. However, since the packet header is large, (e.g., it includes the full source and destination addresses, etc.,) the information text must be large enough to offset the overhead inefficiency introduced by the header.

In the MIXD strategy, voice packets are routed over virtual paths as in the PVC scheme, but data packets are independently routed to avoid congestion points. This is a combination of the fixed path and path-independent approaches, and attempts to associate each routing strategy with the class of data for which it is best suited. Hence, voice packets use the fixed path protocol while data packets use the path-independent protocol. Processing and software are simpler for either of the first two strategies than the combination of MIXD, because all voice, data, and control are handled similarly.

3. Gap Control - Delay has primarily a psychological effect on an ongoing voice call;[15] it has little or no effect on the intelligibility and naturalness of the speech signal, although it is readily detectable and can be very disconcerting. The variability of the delay, however, directly affects the intelligibility of a conversation.[15] When packets arrive at their destination, they will generally exhibit a variation in delay, with respect to the time at which they were sent out, as a result of queuing, nodal processing, and other time varying characteristics. The function of the destination node is to compensate for these variations and to preserve the relative timing of the information when reconstituting the speech for presentation to the listener.

   During transmission, packets may be lost, or may arrive so late that the destination node must discard them. At this point a gap is introduced into the output waveforms that disturbs the rhythm of the speech and therefore impacts intelligibility. Studies of similar phenomenon using a TASI system demonstrated that a gap rate of 0.5% is acceptable to a listener, whereas 2% would be disturbing to the continuity of a conversation.[4] We would like to limit the effect that these gaps have on the ouput voice stream.

   By introducing additional delay, D, at the destination node, (i.e., by buffering the output stream and delaying the first packet of the stream,) the frequency and duration of gaps can be reduced. However, too much delay would destroy the interactive nature of the speech conversation, as well as introduce excessive buffering. Thus, there is a fundamental design trade-off, namely, gap probability verus the delay for

smoothing the variations in packet arrival terms. The delay itself may be selected at the beginning of a sentence based on a sampling of previous delays. Reference 17 discusses several adaptive methods for choosing D.

Unless D is very large, there is a finite probability of a gap occurring (see Figures 3.3-13a through 3.3-13b). Figures 3.3-13c and d show two ways of dealing with this situation.[17] During the experimental phase, we plan to evaluate these schemes and several of their variations with regard to gap prevention or reduction. The initial technique to be used is shown in Figure 3.3-13c. This figure shows that the play-out time of the speech is expanded to include all packets. A late packet causes a delay increment that is added to the overall delay, thus expanding the length of the speech. Buffer length can be designed to accommodate a worst-case accumulated delay. This method emphasizes the delivery of all packets produced by the source node. Timing is considered secondary in that it is necessary only to deliver packets in the same sequence in which they were generated. In this scheme, extremely late out-of-order packets are ignored.

In Figure 3.3-13d a packet (or part of a packet) arriving late is discarded by the destination node. This method preserves timing of the waveform at the expense of throwing away late arriving packets. With this method it is always possible to maintain timing of the speech waveform; however, in general, more gaps will occur than in the previous method.

Combinations of these methods are also possible. For example, the node may discard a packet that arrives while the next successive packet is being output. In general, if packets are late, the time axis is expanded as in Figure 3.3-13c; however, some packets are discarded to maintain reasonable timing. Another procedure is to limit the expansion of the time axis, or the fraction of packets discarded, to a certain threshold, switching from one method to another when the threshold is exceeded.

When a gap occurs, there are various choices to fill the interval prior to playback to a listener. The gap can be left as a silent interval; this would be a good technique for handling long gaps. Information in the previous packet, (e.g., the last frame of the previous packet), can be repeated to fill the gap; this procedure would work well for short gaps. Other sounds that fill the gap and are not detrimental to the listener, (e.g., low background noise), could also be used. Our strategy will be to fill long gaps with silent periods and short gaps with repeated frames. The specific

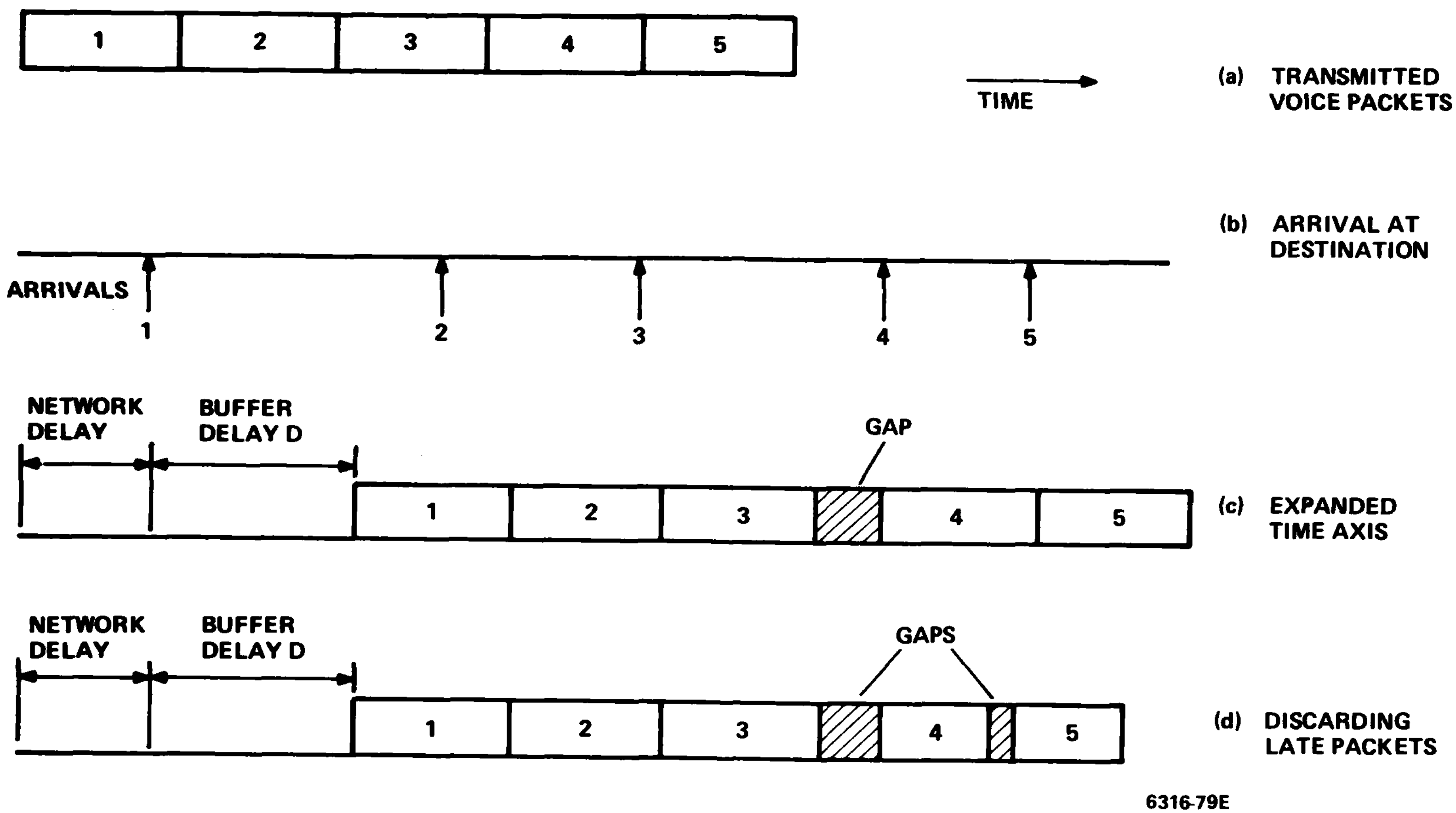


Figure 3.3-13. Playout Methods Used for Gap Control

information used to fill gaps, however, is a function of voice encoding techniques and will be subject to experimentation.18

The introduction of gaps increases the duration of output talkspurts relative to their corresponding inputs. Since this could result in a situation where the delayed output speech lags further and further behind the input, a method must exist for adjusting the pauses between talkspurts to compensate for the increased delay. This transformation of silent intervals into gaps of zero duration will cause the overall delay to become closer to its nominal value.

c. Flexible Hybrid Concept

1. Delay and Voice Continuity - The Flexible Hybrid concept has two different regions and different delay characteristics associated with each of them. The packet (Class II) region displays characteristics similar to the MIXD concept with its variable delays for packets depending on routing and instantaneous loading. The circuit (Class I) region has a more uniform delay that is load independent with very low dispersion. The disadvantage is that this more uniform delay is longer than the average packet delay. The integration of the two regions on one transmission channel has no effect on the circuit region, but does affect the packet region. Delay characteristics for the interactive data are shown in Table 3.3-8. This table compares the mean delay of a PAR packet for the MIXD concept and for the Flexible Hybrid concept. The numbers for MIXD are based on a 65% voice content and are adjusted for percentage of available bandwidth. As can be seen from the data the queue delays for the mean packet are approximately the same for each concept. This is not an unexpected result, considering the bulk content in the MIXD analysis is 20% of load and a 4000 bit packet size, and the circuit region is 20% (3000 bits) of the bandwidth for the Flexible Hybrid approach. It can be seen though, that as the bulk data content grows larger (e.g., to 40%), both concepts would suffer increased mean packet delay. The effect would be more pronounced on the Flexible Hybrid concept, since the 40% bandwidth (6000 bits) would be treated as a single large packet in a queuing model as opposed to individual 4000 bit packets in the MIXD queuing model. Factors not appearing in this analysis are the gains made by prioritizing the PVC voice packets for delivery and the processor loading to handle packet routing. The indications at this time are that with the judicious choice of bulk packet size, the

TABLE 3.3-8. MEAN PACKET DELAY FOR FLEXIBLE HYBRID AND MIXD

Loading Percentage of Available Packet Region	DELAY	
	MIXD	FLEXIBLE HYBRID
25%	0.25 ms	.25 ms
50%	0.40 ms	0.40 ms
75%	1.5 ms	2.0 ms

delays for the MIXD concept are slightly better than the Flexible Hybrid concept, but at the cost of bandwidth efficiency.

2. Variable Frame Size - In order to maximize efficiency and to improve the delivery requirements of packet switched voice and data in the Flexible Hybrid concept, the boundary between succesive frames will be made variable.[11] Thus, the node will attempt to transmit the next frame at a constant time interval, normally every 10 ms. However, the new frame will not be started if a packet is in transmission at that time, but will be transmitted as soon as the packet is completed. An advantage to this method is that if the outgoing link is available when a packet arrives, the packet can be transmitted in the capacity coming just after its arrival, without waiting for the next frame cycle. This results in a decrease in the expected delay for packet switched voice and data. Delay for circuit switched traffic will vary; however, this can be upper-bounded to a maximum value that is acceptable for the transmission of the Bulk, Fax, and Burst traffic transmitted in the circuit switched region of the frame. The additional delay and delay jitter incurred by the circuit switched traffic will have a minimal effect on delivery service.

3. Error and Loss Strategy - The error and loss strategies for the Class II data in the Flexible Hybrid concept are identical to those of the MIXD concept for similar traffic. The error and loss strategy for Bulk and all other data carried in the circuit region will be left to the host equipments. The data will be transported without error control. Since the "circuits" are full-time and synchronous, there is no discard policy other than preemption. The host user equipments will request retransmission as required by their particular needs. Any preemption will be done with notification.

3.3.3.8 Classmarks

In the two concepts to be implemented for the ILRAN network, there are common and similar requirements for classmarking. The three general types of classmarks are: terminal classmarks that are associated with the terminal access portion; call associated classmarks that characterize a call in progress; and the traveling classmarks that are temporarily stored at each switching point as a call travels through the network.

a. MIXD Concept - The classmarks used for the MIXD concept are in many cases a subset of those used for the Flexible Hybrid concept. The variations result from the way a particular type of traffic will be switched. In order to determine the number of classmarks necessary, the different combinations of service will have to be accounted for. The absolute number is also dependent on the implementation of the classmarking tables.

1. Terminal Classmarks - The terminal classmarks are those that are used at the time of assignment of each terminal in the system. They are used to define a terminal's characteristics and privileges. Figure 3.3-14 is an indication of the parameters that must be specified. In addition to these are the implied characteristics of the hardware interfaces and protocols. These would include ring techniques, off-hook detection, seizing, signaling method, and general first-level interface parameters that do not affect the backbone network. Whether these parameters have to be treated as assignable classmarks is dependent on hardware/software implementation of the interfaces.

2. Call Associated Classmarks - At the establishment of each call a temporary classmark is assigned. It characterizes the call to derive routing, preemption, and call establishment procedures. The classmark data is stored at the originating and any other involved nodes to the extent that is required at that node. In a generic sense, the data will be stored in different cross-referenced tables according to the use of the data. The classmark data that is required is shown in Table 3.3-9. The precedences shown in this table are the mapped or derived hierarchical precedences used for traffic handling and preemption.

3. Traveling Classmarks - These classmarks are the variables associated with a call that travel with each packet. These classmarks are used at the different nodes to establish much of the call associated classmark tables. The coded classmark contains the call identifier number, any feature being used, derived precedence, requested routing if any, and security classification. Much of the information comes from the originating node and is a combination of terminal classmark information and call associated information.

b. Flexible Hybrid Concept - The classmarks for the Flexible Hybrid concept are an expansion of the MIXD classmarks. The same three categories of terminal, call associated, and traveling classmarks are required. The only additions are the extra fields necessary for circuit traffic.

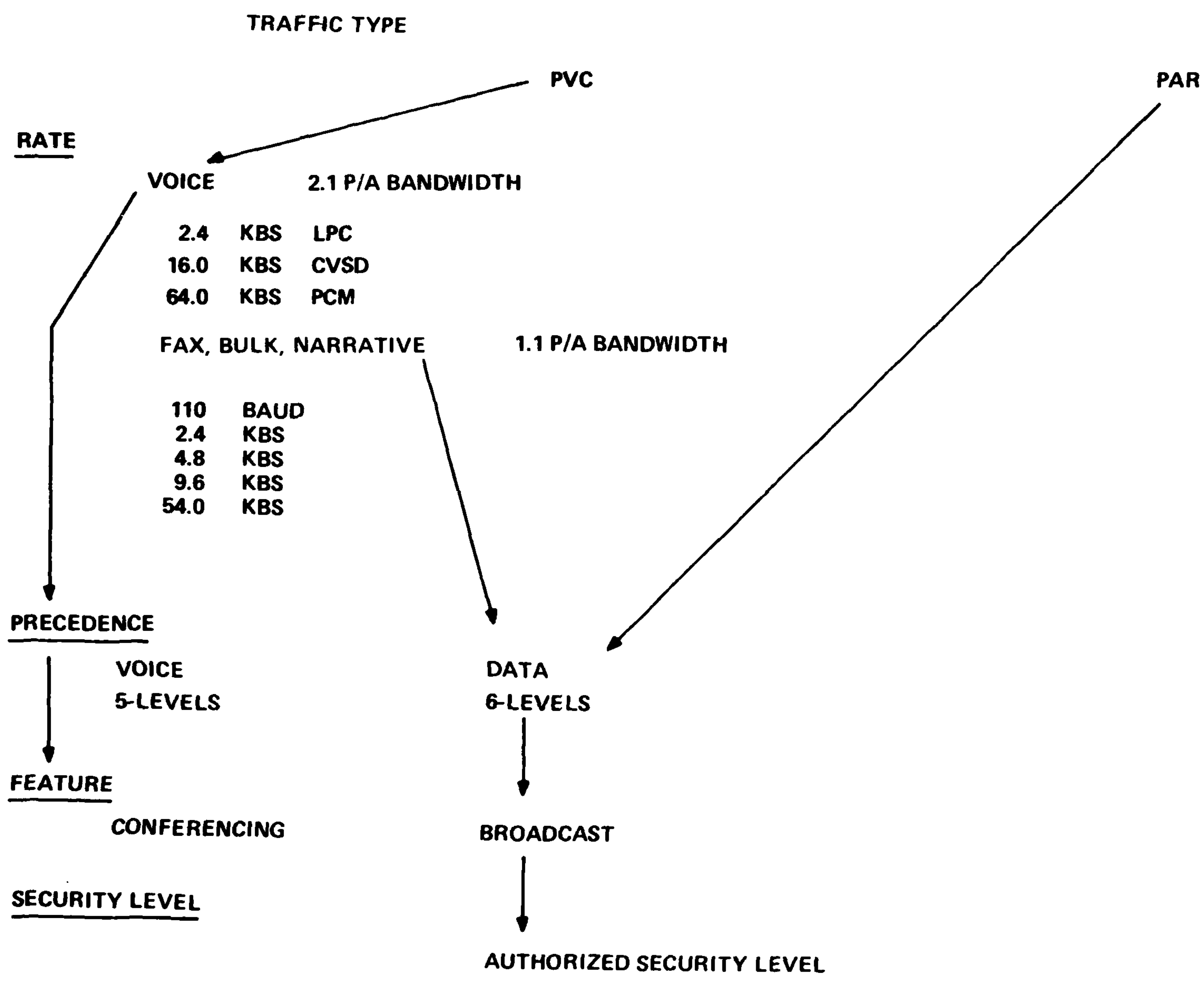


Figure 3.3-14. MIXD Terminal Classmark

TABLE 3.3-9 MIXD CONCEPT CALL ASSOCIATED CLASSMARK

Precedence	- 6 to 10 categories of derived precedence mapped from dialed precedence level of the call and type of data being transmitted. The order of preemption may be different from the order of transmission queue. (See precedence and preemption sections.)
Call Type	- Stored by call identification number to indicate originating, terminating or tandem call.
Route	- Stored by call identification number to indicate the trunk being used. This information is derived from the routing algorithm or from information in the traveling classmark indicating a preselected route.
Feature Indicator	- Stored by call identification number to indicate that a call is a broadcast data or voice conference call. The information is used mainly at the originating node and is cross referenced to the parties of the conference.
Security Classification	- Stored by call identification to indicate the security level established for the call.
Switching Type	- Stored by call identification number to indicate how the packet is to be handled. (PAR, PVC)

1. Terminal Classmarks - The terminal classmarks for the Flexible Hybrid are an expansion of those for the MIXD concept. Figure 3.3-15 shows the classmark requirements for the Flexible Hybrid concept.
2. Call Associated Classmarks - Call associated classmarks for the packet region in the Flexible Hybrid concept indicate the same characteristics as the classmarks in the MIXD concept. The addition of the circuit region does add the requirement for more information to be stored to distinguish between the different regions. Table 3.3-10 shows the different characteristics that will be stored for call associated classmarks in the Flexible Hybrid.
3. Traveling Classmarks - The traveling classmarks for the MIXD and the Flexible Hybrid concepts have to indicate the same information as that used in packet calls. In the circuit region the information as to call security, precedence, etc., is also required, but it is transmitted by CCIS messages instead of trailblazer packets as in PVC.

3.3.3.9 Service Features

Examination of service features as an evaluation criterion is intended to determine those features that may have an impact on the ultimate choice of a voice/data integration concept for the future DCS. Service features can be defined as those services offered by a telecommunication network that increase the efficiency of the subscriber or the network, or in many cases, both. Service features, for purposes of this investigation, do not include line or trunk interfaces, ordinary call progress tones, or numbering plan flexibility. A list of service features considered in this examination is given in the accompanying Table 3.3-11. The list is intended to be representative of military network service features and is not intended to be exhaustive.

The results of our investigation show that there are two subscriber service features whose performance may depend on the voice/data integration concept employed. The features are:

a. Voice Conferencing
b. Data Multiaddressing.

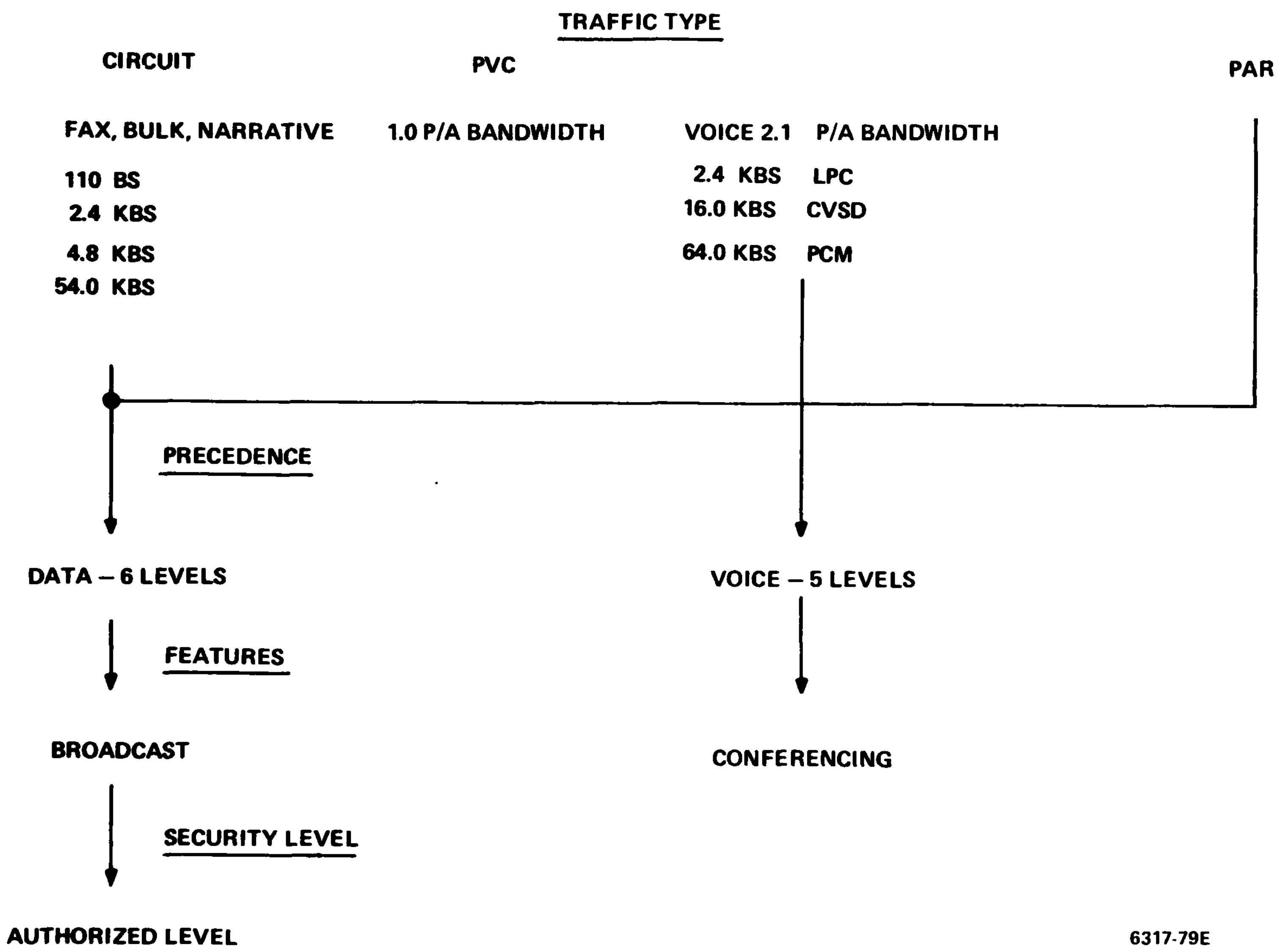


Figure 3.3-15. Flexible Hybrid Terminal Classmark

TABLE 3.3-10. FLEXIBLE HYBRID CONCEPT

CALL ASSOCIATED CLASSMARKS

Precedence	- 6 to 10 categories mapped from the dialed precedence level of the call and the type of data and switching technique being used. The precedence is used in resolving the overall priority for bandwidth allocation between the different regions. A subset of these priorities is used to form the queues for packet transmission.
Call Type	- Stored by call identification number to indicate originating, terminating, or tandem call. In the circuit region, the call identification number is derived from the circuit region map.
Route	- Stored by call identification number to indicate the trunk being used. The information is determined by the routing algorithm of the node or by requested route information in the traveling classmark. In the case of the circuit region it also indicates the time slot identifier in the circuit region.
Feature Indicator	- The classmarking for features is independent of concept and is identical to that of the MIXD.
Security Classification	- The classmarking for security level being used is independent of concept and is identical to that of the MIXD.

TABLE 3.3-11 REPRESENTATIVE SERVICE FEATURES

Abbreviated Keying ("speed calling")
Archival Storage
Automatic Call Forwarding
Call Tracing
Calling Number Display
Camp-on Busy (including call waiting signal and busy-line callback)
Consultation Hold and Call Transfer
Data Multi-Addressing
Dictation/Word Processing Access
Direct Inward and Outward Dialing (DID/DOD)
Electronic Mailbox
Line Hunting Groups
Message Retrieval
Message Tracing
Paging Access
Recorded Announcements
Remote Call Pickup
Switched Hot Line Service
Unanswered Call Diversion
Voice Conferencing (random, preset, "meet-me", etc.)

We concluded that all other commonly accepted service features appear indifferent to the integration concept, since feature use is signaling related, and packet communications are used for control in both hybrid and packet integration schemes.

a. Voice Conferencing - Dependence of voice conferencing on the integration concept occurs not because of the link load, which could be simulated by several two-party conversations, but because of the need for synchronism among speakers during a conference. It is possible that the time synchronization of speakers may be more complex in an asynchronous voice packet system with its greater dispersion and delays. Because of the importance of conferencing in the DCS, and because the next several years should see more techniques proposed in true digital voice conferencing, it seems wise to incorporate a multiple-speaker digital conference capability in the TSN for evalution of conference usefulness and quality as functions of the integration concepts under study.

It should be noted that voice conferencing as a communication technique will not be under examination, since this process has long been implemented in analog systems, and many other organizations are currently studying approaches to effective digital conferencing. It is our objective to incorporate in the TSN a conference capability that has the flexibility to be used with both hybrid and packet integration concepts at the least cost consistent with system objectives.

Because the integration concepts to be evaluated in the TSN will be implemented in the digital area of DCS III, it is important that TSN voice conferencing be digital in nature, and this is the approach that we recommend. While it would be possible to use analog bridges employing D/A conversion, analog summation, and A/D conversion, that approach introduces the undesirable analog effects of instability, loss, and impedance mismatching that could obscure the results of integration concept testing.

The two approaches to voice conferencing appear appropriate - voice control and panel control.[19] These are discussed in the following sections.

1. Voice-Controlled Conferencing - The digital conferencing technique to be employed in ILRAN nodes will use a voice controlled signal switching approach[20] as opposed to mixing or "bridging" of signals as in an analog approach. In this "instant-speaker" algorithm, encoded voice samples from all conferees are sampled within a constant nominal time interval. The channel with the highest amplitude is selected as the speaker, and his voice samples are transmitted to the other conferees,

but not back to himself. The voice sample with the next largest amplitude (above a minimum activity threshold) is sent to the current speaker under the assumption that the interrupt process is a primary mode for interactive communications.[21] The digital conference facility is sketched diagrammatically in Figure 3.3-16.

Digital conferencing schemes similar to the above have been studied by GTE[20,22] and by researchers at MIT Lincoln Laboratory.[23]

For maximum simplicity consistent with experimental objectives, only 64Kbps PCM voice subscribers will be allowed access to conference facilities. This will permit satisfactory testing of the effect of the integration concept on voice conferences, yet eliminate the need to provide a similar facility for any other voice digitization algorithm (e.g., CVSD) in the TSN.

2. Panel-Controlled Conferencing - In panel-controlled conferencing, each conferee is provide with a special control panel that allows him to exchange control messages with the conference chairman. By the use of the panel, each participant requests permission to speak or to interrupt another speaker or notifies the chairman that he has finished speaking. The chairman uses his panel to permit or to rescind a conferee's right to speak and to notify them of speaker changes. Control messages to accomplish all of this are exchanged between the control panels over the communications network.

   Note that in this model all participants listen to only one source at a time under control of the chairman who gives the right to speak to a conferee at the proper time. Distribution of conferee voice segments or packets can be made by concentrating these at the chairman's mode and redistributing them from there to all participants, or by sending the packets directly from the originating source to all other participants. Direct distribution requires a somewhat more complicated control scheme because the voice paths are not the same as the control message paths, but it probably achieves a more efficient conference interchange.

   An overview of the network for teleconferencing and control connections is shown in Figure 3.3-17. Note that the chairman can be either a person or an automatic program.[28] It should be added that an advanced conferencing capability could include transmission of data messages as well as voice communications among participants. In that case the conference techniques may be even more dependent upon the voice/data integration techniques that are employed.

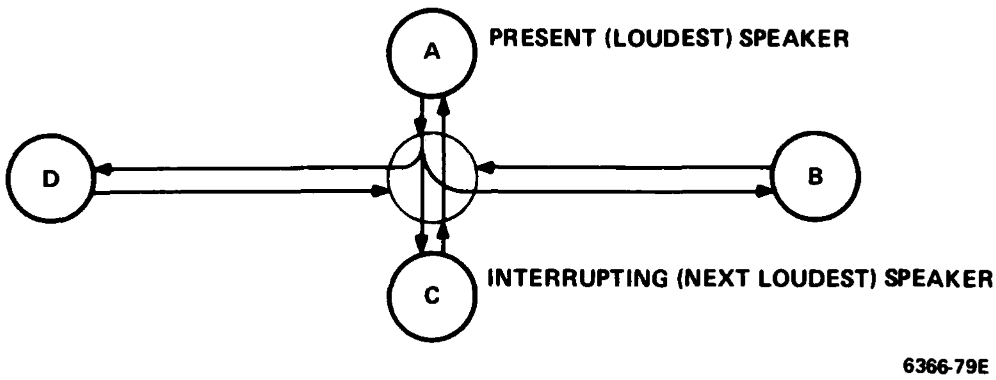


Figure 3.3-16. Digital Conference

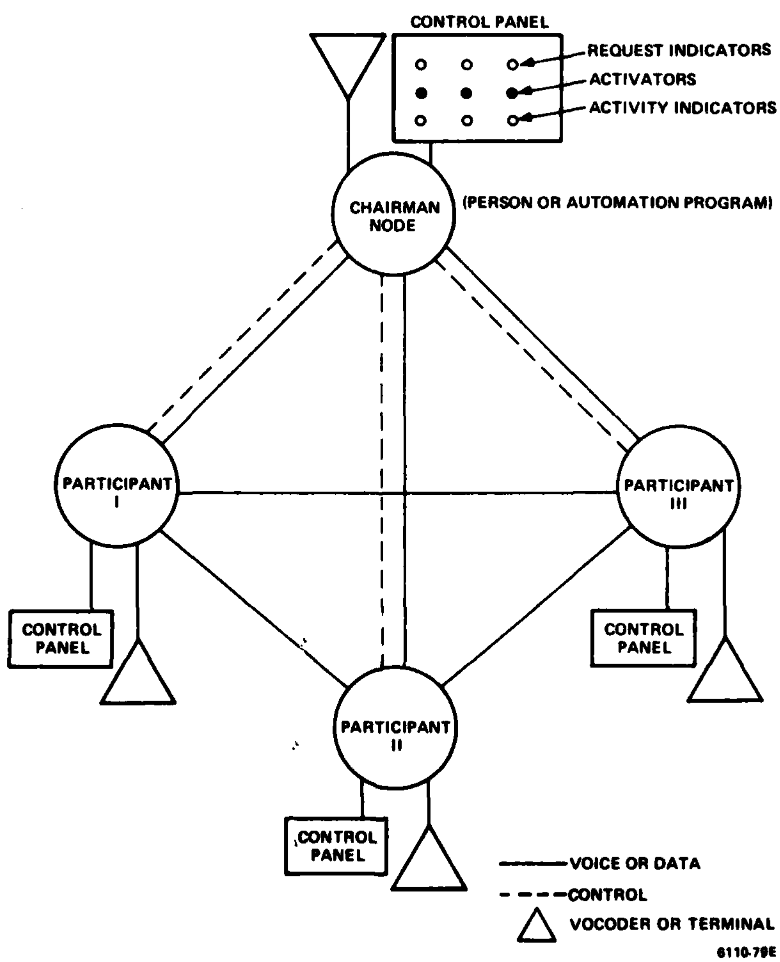


Figure 3.3-17. Data and Control Connections for Teleconferencing

Whereas voice-controlled conferencing requires only ordinary speaker instruments and amplitude detection hardware in addition to the controlling software, panel controlled conferencing requires development of the control panels and their operating protocols. The higher cost of the panel controlled approach will preclude it from being incorporated initially in the TSN, although it will undoubtedly be of interest at some later date.

b. Data Multiaddressing - Multiaddressing is normally considered a data traffic service. It provides transfer of data messages from one source to many destination nodes through the network. Conceptually, there are two methods in which the distribution of messages to the addressees may be accomplished. Forwarding would pass each multiaddressed packet successively from one node to the next, through all the nodes addressed, by setting up pertinent forwarding tables in each of these nodes. Multiaddressing by broadcasting dispatches as many duplicate messages as there are addresses. A direct virtual connection is established between the source and each addressee as illustrated in Figure 3.3-18. In a large network, a single packet will probably reach every addressee faster by broadcasting than by forwarding, and comparison of the two techniques would be a suitable subject for testing under different integration concepts. However, in a network of only a few nodes, such as the TSN, there does not appear to be any significant difference between the two multiaddressing techniques. Therefore, broadcasting has been chosen as the preferred method for the transmission of packets from one source to many destinations under multiaddressing.

It should be noted that the multiaddressing function is itself independent of concept, since it will be performed by adaptive routing of data packets under any hybrid or "pure" packet integration approach. The effect of multiaddressed messages on voice, interactive data, or other real-time or near-real-time communications will be the proper subject for experimentation.

3.4 NEARER TERM AND TRANSITIONAL INTEGRATED SWITCHING CONCEPTS

The integrated concepts described in sections 3.2 and 3.3 have long range objectives whose realization is based on development of a new generation of communication switching systems. These future systems will take advantage of advanced packet or hybrid circuit and packet switching techniques and digital VLSI technology to provide voice and data communications in an integrated architecture, as well as provide additional features, such as electronic mail and computer

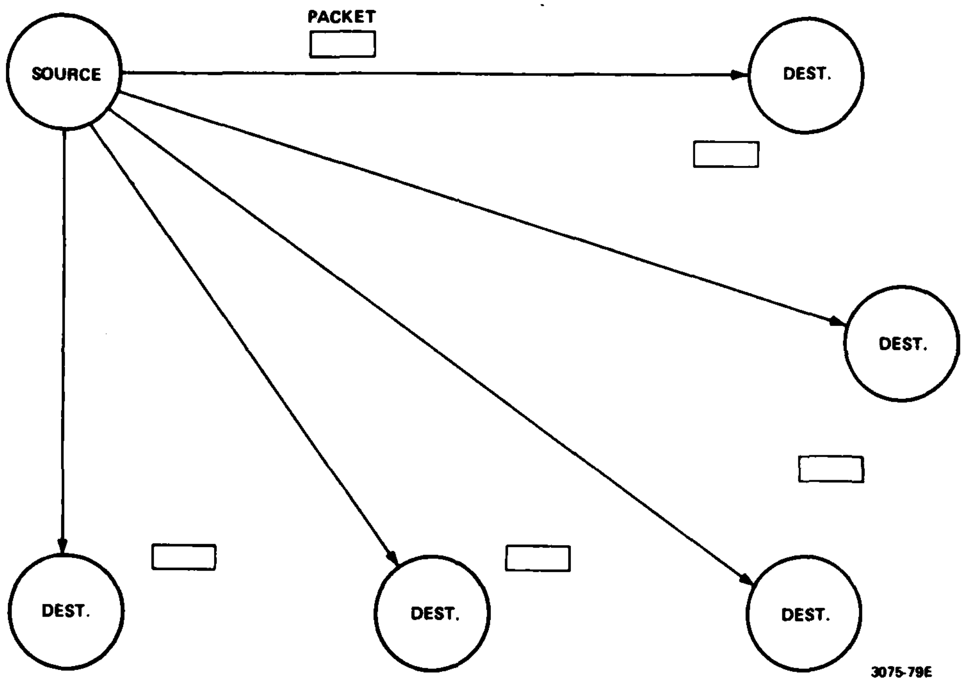


Figure 3.3-18. Multiaddressing by Broadcasting

teleconferencing. In addition to these long range objectives, however, there are short range objectives for providing advanced features and networking capabilities in current military networks. These short-term objectives encompass the realization of data switching, common channel signaling, advanced network control, and digital speech interpolation, and are best realized by incorporating certain key capabilities in an existing family of digital switches or PABXs.

There are several strategies for integrating voice and data through the use of existing equipments in order to meet these short-term objectives. For example, packet switched subscribers could be serviced in an already existing circuit switched network by requiring packet switches to access circuit switches for desired bandwidth. Allocation of bandwidth by packet switched subscribers would be conducted according to circuit switched network protocol, i.e., requests for additional capacity for packet switched transmission would be "dialed-up". However, once transmission capacity is allocated, the information transfer would be carried out using packet switching techniques. Another strategy is to have existing circuit and packet switches gain low level access to the network through a controller which shares allocation of transmission capacity among both circuit and packet switched subscribers. Each switch obtains bandwidth allocation for its voice or data transaction according to its own unique set of protocols. Both these strategies provide the advantage of easing the transition to an integrated network capable of handling diverse traffic types while making maximum use of existing facilities.

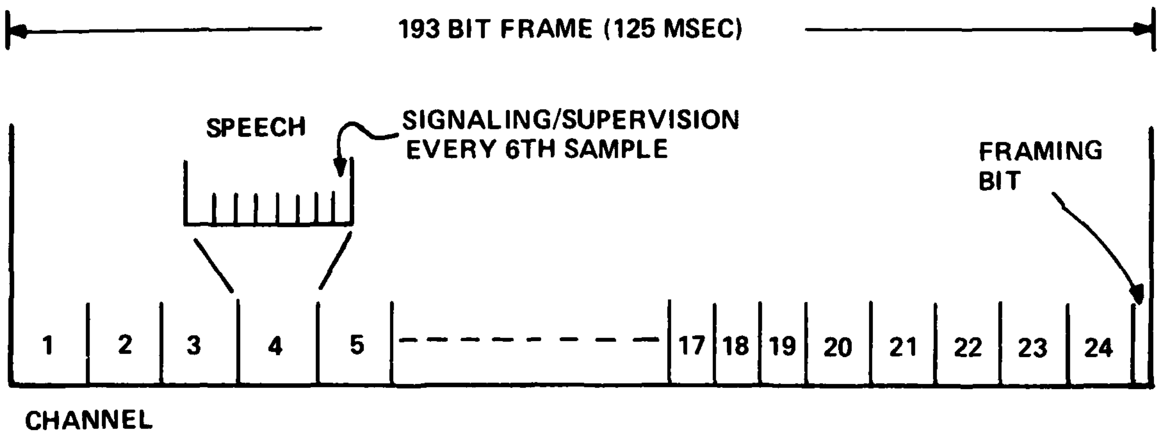
3.4.1 Integration of Voice and Data Traffic on T1 Link

The rapid growth of computers in the military environment as well as in industrial and government organization, has resulted in an increased requirement for data communication facilities to interconnect widely distributed computer and terminal equipments. Examples of this include access to centralized data bases by data terminals, terminal-to-terminal communications, and host-to-host communications. Typically, these data communications needs are serviced by multiple networks (separate from voice), dedicated to a particular application

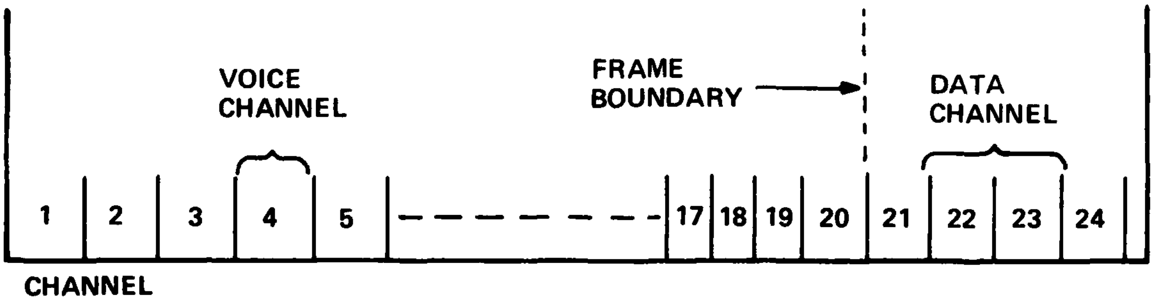
and administered by separate organizations. The proliferation of separate, compatible facilities and networks results in inefficiencies due to duplication of equipment (and organizations for network management) as well as inefficient utilization of leased transmission facilities. An integrated voice and data network, which provides a capability for communicating between incompatible terminals and hosts, allows sharing of transmission facilities for voice and data, and is an attractive alternative to the separate and dedicated facilities which exist today.

Figure 3.4-1a illustrates the format for a time-division-multiplexed (TDM) Bell T1 frame structure, in which a predetermined portion of the transmission link is dedicated to continuous use by a voice (or data) subscriber for the duration of the call. For the T1 carrier, provision is made for 24 64Kbps PCM speech channels, multiplexed for transmission at a T1 rate of 1.544 Mb/s. Individual channels contain 8 bits each and repeat at a basic frame rate of 125 μs. Every 6th frame, the 8th bit of a channel is used for in-band supervision and signaling, in order to setup and breakdown individual calls. The last bit in each frame, the 193rd bit, is used for establishing and maintaining synchronization.

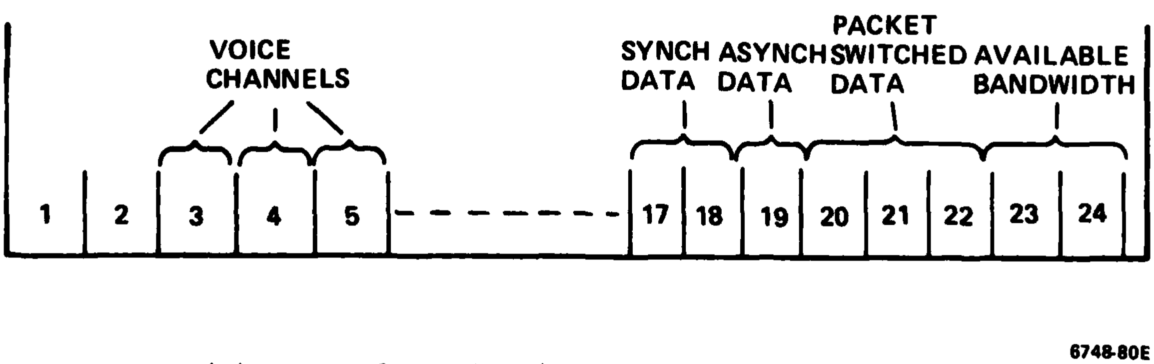
A straightforward way of handling voice and data in the same system is to time-share the network to provide data transfer during idle periods, e.g., voice facilities could be used at night for batch transmission. However, by proper assignment of channels, voice and data can be integrated concurrently on the same transmission link. Certain channels could be "nailed" down, a technique in which a non-switched connection is made to a predetermined T1 channel, and used for 56 Kb/s data transmission. Figure 3.4-1b shows an example where contiguous T1 channels are grouped together to form separate voice and data regions. The boundary between regions may be fixed or could vary with traffic. Thus, during idle periods, traffic assigned to one region could make use of unused capacity nominally assigned to the other region. Figure 3.4-1c shows an interleaved approach in which channels (individually and in groups) are assigned to voice or data subscribers on demand.



(a) Bell T1 Carrier PCM Frame Structure



(b) Separate Voice & Data Regions



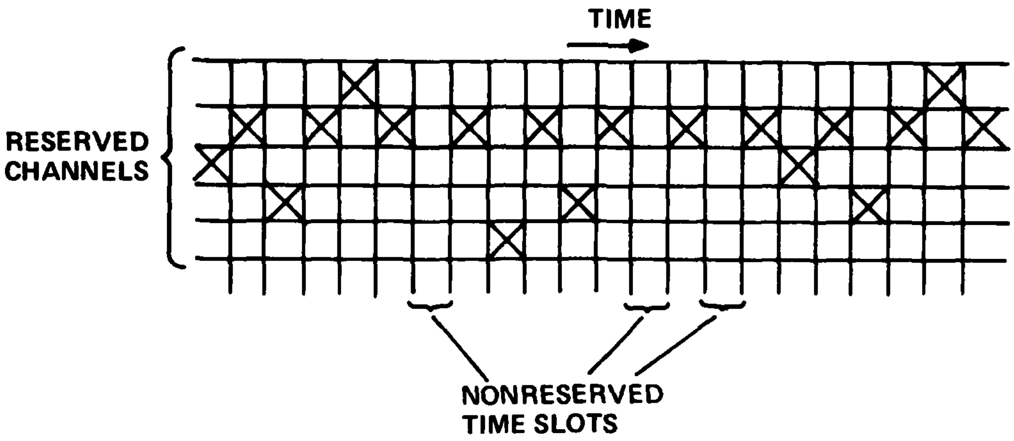
(c) Interleaved Voice and Data Channels

Figure 3.4-1. TDM Channel Allocation Strategies for Combining Voice and Data on a T1 Link

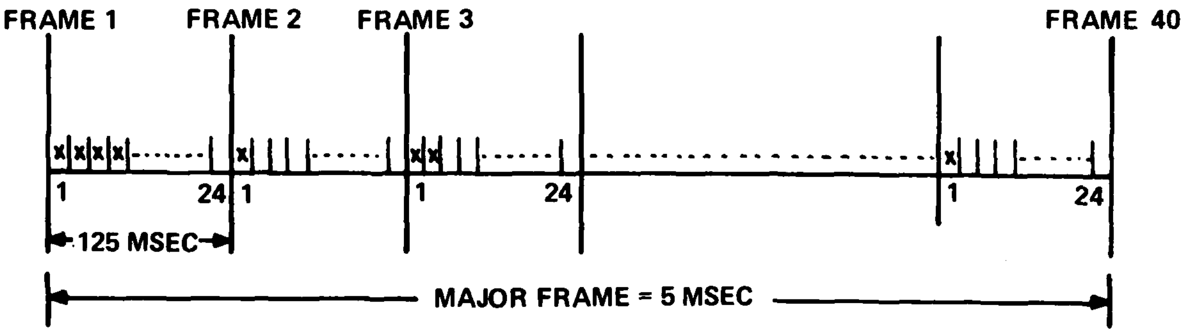
In the examples given above, channels are first reserved prior to transmission. Channels thus allocated cannot be used for any other purpose unless they are disconnected. This type of transmission is very suitable for synchronous traffic, such as voice, facsimile, or long data files, which require continuous transmission, since fragments of these calls can fit nicely into preassigned time slots. The same technique can be used to carry asynchronous traffic, e.g., packet switched traffic such as interactive data between a terminal and computer, where the traffic is not continuous, but occurs according to a nonregular pattern in time. This type of data is "bursty", i.e., it exhibits a high peak to average transmission ratio. Allocating entire channel widths to these types of calls for the full duration of the connection can be very wasteful of bandwidth, particularly for the case of asynchronous, low speed subscribers.

A form of switching which combines synchronous and asynchronous time division together in the same system in a dynamic fashion would help to alleviate this problem.[25] The basic principle is illustrated in Figure 3.4-2a. Time slots are first assigned to continuous (e.g., voice), interactive, or batch-type data according to an established priority. All remaining time slots then become eligible for nonreserved transmission (e.g., to carry packet switched data). Nonreserved time slots can be treated separately, combined into channels of different widths, or treated as a single large channel of variable width. Interactive or real-time data would require small channel widths whereas batch or nontime dependent data would require large channel widths. Each unreserved time slot must carry addressing and control information with it. Storage is needed to queue the nonreserved traffic when time slots are not immediately available to carry it.

Different voice and synchronous data transmission rates can be serviced concurrently using this technique. In Figure 3.4-1b, the 64 Kb/s voice call requires the reserving of one time slot per frame period or 40 time slots during the 5 msec. major frame. The 9.6 Kb/s call requires the reserving of one time slot every 5 frames or 8 slots



(a) Time Slots Available for Nonreserved Transmission



TRANSMISSION RATE KB/S	NO. OF RESERVED SLOTS PER 5 MSEC MAJOR FRAME PERIOD	NO. OF BITS PER BYTE NEEDED	CHANNEL ASSIGNMENT
64	40	8 OUT OF 8	1
56	40	7 OUT OF 8	–
32	20	8 OUT OF 8	2
16	10	8 OUT OF 8	3
9.6	8	8 OUT OF 8	4
4.8	4	8 OUT OF 8	–
2.4	2	8 OUT OF 8	–
1.2	1	6 OUT OF 8	–

6754-80E

(b) Variable Transmission Rates for Reserved Transmissions

Figure 3.4-2. Combined Synchronous and Asynchronous Time Division

per major frame. It is also possible to handle a 56 Kb/s data stream in an identical manner to the 64 Kb/s voice call by using 7 out of every 8 bits to transmit the information.

Packet switching can be looked upon as a form of asynchronous time division switching in which each fragment of the call contains its own addressing and control information. For certain applications, e.g., interactive data, packet switching is much more efficient than techniques which allocate continuous fixed capacity channels to each user. In addition, it provides a powerful error checking and retransmission capability. However, the methods by which packets are formed and transmitted through a network can often introduce significant amounts of transmission delay and overhead, particularly for low speed terminal traffic. The PACUIT scheme[26] is a method for minimizing the end-to-end delay and improving the transmission efficiency by combining certain aspects of both packet as well as circuit switching. In the PACUIT system, a composite packet is formed, composed of data from multiple users, all traveling between the same source and destination (see Figure 3.4-3). These composite packets are transmitted at regular intervals and contain only information from user terminals which have data to send at the time of transmission. This is in direct contrast to standard packet switching systems where packets are held until a minimum amount of data has been accumulated. A proper choice of transmission rate helps to insure a healthy balance between low buffer size requirements and overhead.

Composite packets are then circuit switched through intermediate nodes of the network without disassembly/reassembly of the constituent packets. Along the path, packets are neither error checked nor stored; error checking and packet control functions are performed only at the source and destination nodes. Overall, the effect is to reduce the end-to-end delay as well as to reduce the buffer size requirements at each node. The technique also proves to be more efficient for handling asynchronous, low speed users than standard packet switching techniques since it permits the transfer of small messages or portions of longer messages, without paying the price of high packet overhead.

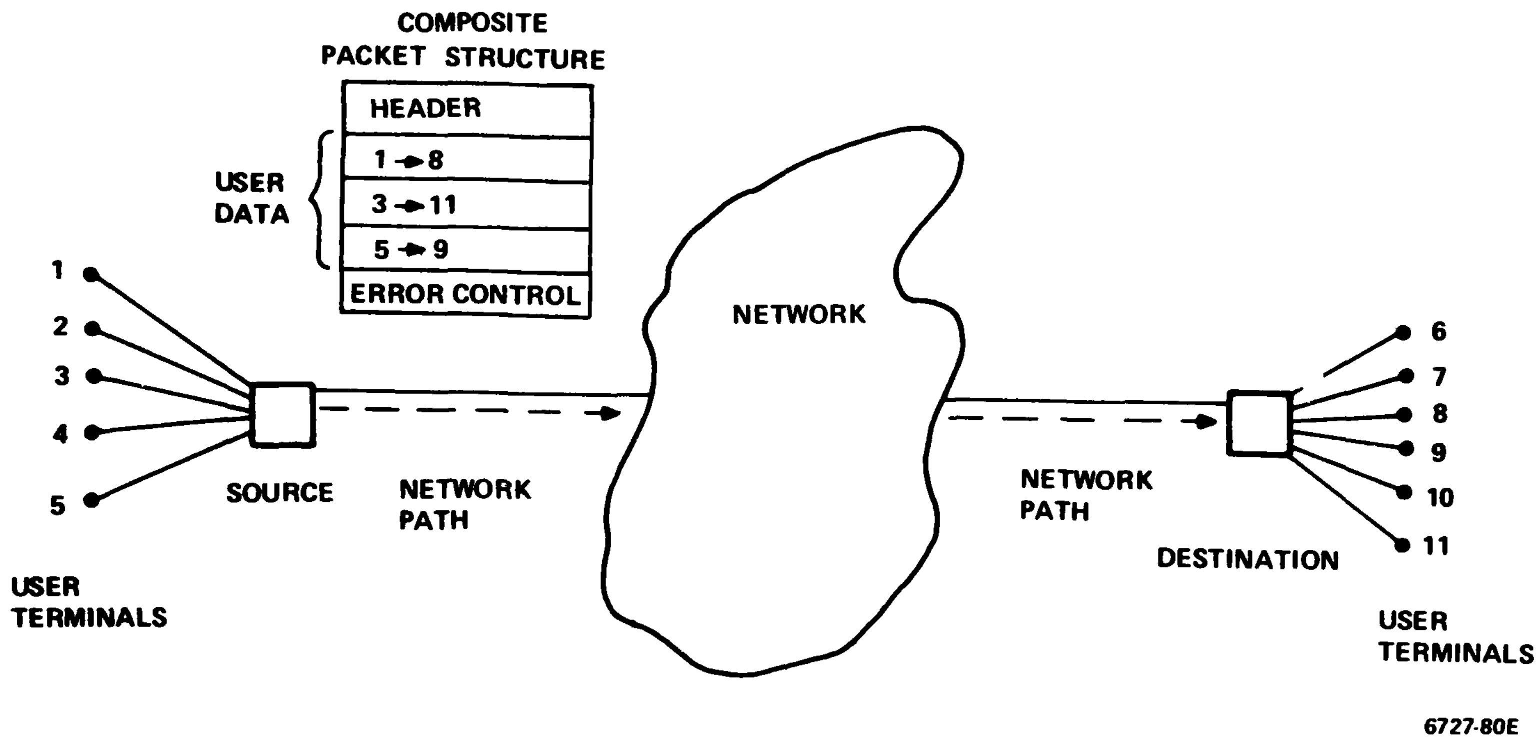


Figure 3.4-3. Example of PACUIT Technique

3.4.2 An Example of an Integrated Network Architecture

Figure 3.4-4 shows a network architecture capable of supporting the integration of voice and data on a T1 link, resulting in a more efficient use of transmission facilities. In this example, circuit switches serve the function of local switching offices as well as tandem switching nodes in the network. These communicate with packet switches in order to establish data connections, receive/transmit internode signaling messages, and provide for interface to the netowrk monitoring/control processor. Local data calls between compatible terminals are handled by the individual circuit or packet switch. For network data connections, circuit and packet switches communicate with each other, passing address and control information needed to setup a path. Common channel signaling is then used to setup a path through the network.

On the T1 link, 22 channels are reserved for standard PCM connections while the remaining 2 channels are fixed for insertion of 56 Kb/s data streams from the packet switch into the T1 bit stream. This can be achieved either by modifying the T1 interface of an existing circuit switch or using an external multiplexer or microprocessor based configuration which would perform the tasks of buffering, synchronization and insertion of the data bit stream into fixed channels of the PCM T1 bit stream.

The packet switch would interface a variety of terminals, concentrators and multiplexers at speeds of up to 56 Kbp/s. Terminals with high utilization which require a dedicated port and host computers implementing file transfers are examples of equipment which would interface directly to the packet switch. In the latter case, high speed internode file transfer between host computers can take place during off hours, resulting in further savings due to high utiization of transmission facilities. A data subscriber at the circuit switch (with the proper classmark) can communicate with a host computer through low speed data lines between circuit and packet switches setup for this purpose. The packet switch performs code, speed and format conversion needed to interconnect incompatible terminals and hosts, and allows high speed access (56 Kbp/s) to offnet data bases in public

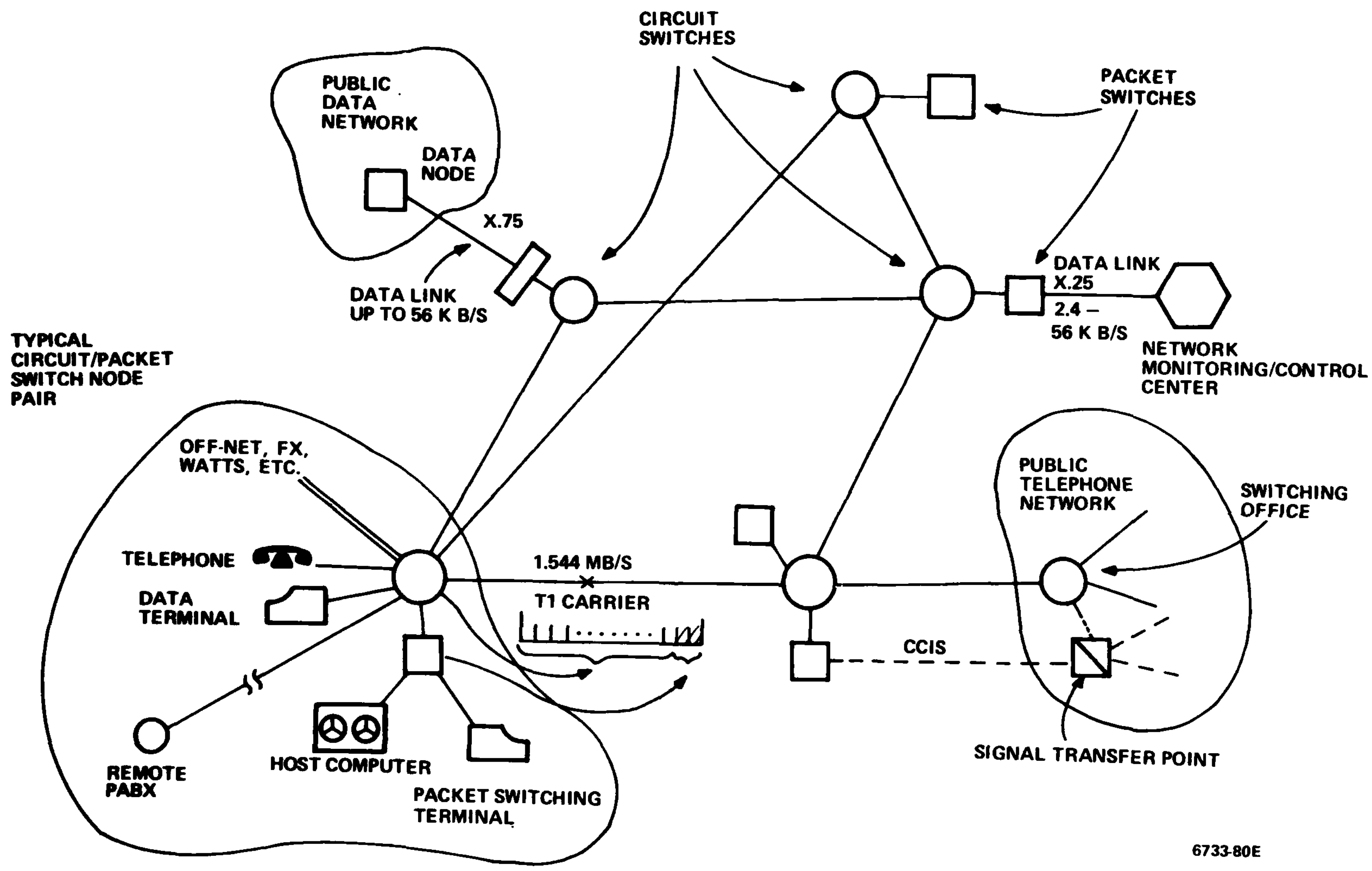


Figure 3.4-4. Example of an Integrated Voice/Data Network

data networks (e.g., Telenet, Tymnet, etc.), using X.25 and X.75 network protocol. Furthermore, it has the capability to multiplex a number of low speed data input streams into one or more high speed (up to 56Kbps) data streams for transmission to another data switching node.

3.4.3 Methods of Improving Performance

There are various methods of improving the performance of integrated systems based on the use of existing equipments and techniques. In Common Channel Signaling (CCS), signals pertaining to the trunks interconnecting switches are transmitted on a separate out-of-band signaling link. As a result of this, certain advantages become evident:

a. A lower call setup time because of the increased signaling speed,

b. Immunity from talkoff problems resulting in lower signaling errors,

c. Improved conversational occupancy of network trunks through elimination of signaling occupancy on those trunks, and a look-ahead capability which permits the originating subscriber to be informed of the called party status by the outgoing exchange,

d. The capability to carry maintenance, administration, and control information for the centralized management of networks, and the introduction of network features and services not possible with in-band signaling systems.

Currently, there exists two major common channel signaling schemes:

a. CCITT No. 6 and its variations (the Bell system's CCIS, used in the North American Telephone Network and Federal Standard No. 6, a militarized version of CCITT No. 6), optimized for an analog network environment,

b. CCITT No. 7, optimized for use with digital networks, and exhibiting similarities to X.25 data protocol.

A comparison of CCITT No. 6 with CCITT No. 7 is presented in Table 3.4-1. Since there is little structural difference between CCITT No. 6 and CCIS, this comparison is also considered valid between CCIS and CCITT No. 7.

TABLE 3.4-1. COMPARISON CCITT NO. 6 AND CCITT NO. 7

Signaling System Parameters	CCITT No. 6	CCITT No. 7	Comments
1. Signaling Channel	Analog or digital	Digital (see comments)	CCITT No. 7: Provision exists for operation within analog network using modem and bit rates of 4.8 KB or 9.6 KB/sec
2. Signal Speed	2400 bits per sec	Optimized for 64 KB/sec. Usable at lower rates	Delay objectives of No. 6 cannot be met at lower than 19.2 KB/sec
3. Signaling Delay (see comments)	Delay / Answer / Non-Priority: Avg PR: 12 msec / 25 msec Cross-Office: 40 / 65	CCITT No. 6 values provide upper limit	Avg. PR = average processing delay at 2400 bits per sec
4. Signal Priorities	Necessary	Not necessary at speed >19.2 KB/sec	
5. Mode	Associated, quasi-associated non-associated	Associated, quasi-associated, non-associated, superiority over no. 6 on non-associated mode	
6. Error Check Polynomial	8-bit primitive polynomial	16-bit primitive polynomial	16-Bit Polynomial same as in X.25
7. Error Correction	Error correction by retransmission	Two methods: (a) Retransmission on short links (b) Forward error correction on long links	Forward error correction based on preventive cyclic retransmission
8. Signaling Dependability	Undetected Error $\leq 10^8$ correct signal units	Undetected signal unit error $\leq 10^8$ signal units	
9. Signal Unit	28 Bits	Signal unit of variable length delimited by flag 01111110.	

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TABLE 3.4-1. COMPARISON OF CCITT NO. 6 AND CCITT NO. 7 (Continued)

Signaling System Parameter	CCITT No. 6	CCITT No. 7	Comments
10. Label	11 bits for CCITT No. 6 13 bits for CCIS (EAX No. 3)	40 bits (14-bits DPC 14 bits OPC and 12 bits CIC)	DPC: Destination code OPC: Origination code CIC: Circuit identification code
11. Continuity Check	Required	No check required for digital speech paths. Check provided for analog/mixed environment.	
12. Signal Set	CCITT No. 6 as reference.	Same except for multiple clearback and re-answer signals listed below	
13. Numbering of Multiple Clearback Signals	Reqd (Clearback 1 Clearback 2, etc.)	Signals eliminated	
14. Numbering of Multiple Reanswer	Reqd Re-Answer 1 Re-Answer 2 Re-Answer 3	Numbering not req'd. Multiple signals replaced by re-answer	
15. Reasonableness Checks	Required at exchange processor	Not required Terminal discards invalid signals	

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In order to take advantage of the benefits of common channel signaling, the relative merits of both signaling schemes should be investigated for possible implementation in a near-term integrated network. The integration of signaling traffic with data traffic for internode data communication, in order to reduce costs by sharing common transmission facilities, should also be considered.

Time Assigned Speech Interpolation (TASI) is a technique (covered in Section 3.2) for overcoming the poor transmission efficiency associated with the traditional circuit switching of voice (see Figure 3.4-5). In the case of voice, the channel is only in use approximately 40% of the time. With a sufficiently large number of channels, most of the capacity of the transmission link can be filled, giving a significant "enhancement" in transmission capacity. Another method used to improve efficiency is to submultiplex voice (e.g., via vocoder techniques) or data channels to service a large number of subscribers. Individual subchannels could be used to carry voice, synchronous or asynchronous data, or facsimile.

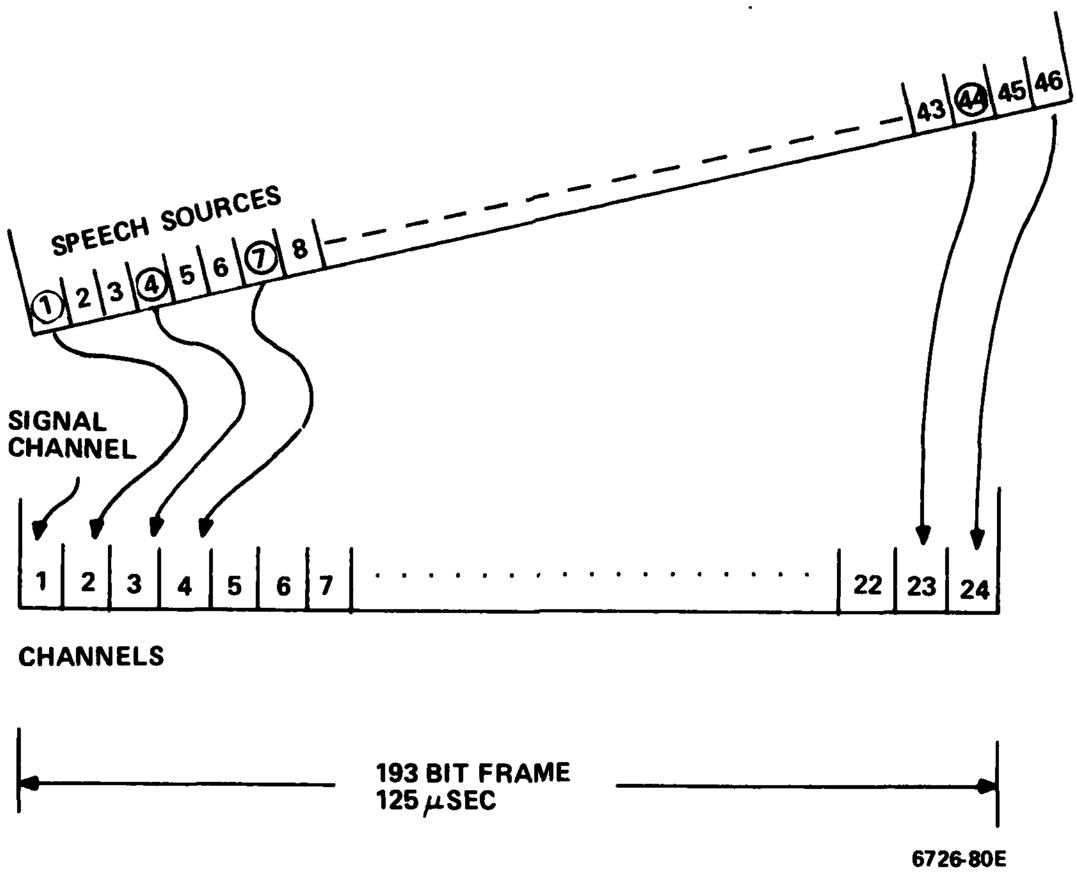

Figure 3.4-5. Use of TASI to Carry Additional Voice Conversations

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SECTION IV

EXPERIMENTS

4.1 INTRODUCTION

Section II identified key nearer term/transitional and far term issues which are seen as appropriate candidates for examination by means of an experimental test bed. This section identifies and describes experiments which are recommended for addressing those issues. These experiments drive the test bed attributes and requirements which are covered in Section V of this report. Section II examines several candidate test bed architectures which meet the requirements for performing the various experiments.

The recommended experiments fall into eight classes:

I Network Interface Experiments
II Routing Experiments
III Combined Voice & Data Experiments
IV Common Channel Signaling Experiments
V Network Control and System Management Experiments
VI Advanced Voice Experiments
VII Data Experiments
VIII Hybrid Experiments

Each class of experiments consists of one or more groups of experiments, for a total of seventeen groups in all. Figure 4-1 shows, in matrix form, the experiment classes and groups vs. the eight issues identified in the previous chapter, with an indication of which issues are addressed by each experiment group. The numbers in the last column indicate the relative order (according to the criteria outlined in Seciton I) in which the group of experiments are to be performed. Note that each group of experiments comprises one or more individual tests for a collective total of approximately seventy individual tests.

Experiments: Class of Experiments	Experiments: Experiment Group	Issues: Inter-Operability	Survivability	Cost	Transition Strategies	Syscon/Net Mgt.	Security	Operational Capability	Integrated System Concepts	RELATIVE ORDER
I Network Interfaces	Voice Net Interfaces	Xp	Xs	Xs	Xs			Xs		1
II Routing Experiments	Alternate Routing	Xp	Xp	Xs	Xs		Xs	Xs		2
	Tandem Routing	Xs	Xp	Xp	Xs		Xs	Xs		3
	Packet Routing		Xp	Xp			Xs			4
	Flexible Trunk		Xp	Xs					Xs	6
III Combined Voice and Data	T1 Voice/Data				Xs			Xs	Xp	5
	Comb. Voice/Data	Xs			Xs	Xs		Xs	Xp	12
IV CCS	CCS Formats	Xs	Xs	Xs	Xs	Xp				9
	CCS and Features	Xs	Xs	Xs	Xs	Xp		Xp		10
V Network Control	External Monitoring			Xs		Xp				14
	Integrated Control		Xs	Xs	Xs	Xp				11
	Hybrid Net. Control		Xs		Xs	Xp			Xs	13
VI Voice Experiments	TASI			Xp	Xs		Xs			7
	Packetized Voice			Xp	Xp		Xs		Xs	8
VII Data Experiments	Data Experiments	Xs			Xs		Xs	Xp		15
VIII Hybrid Experiments	Fixed Bandwidth Variable Boundary			Xs	Xs				Xp	16
	Variable Frame Variable Boundary			Xs	Xs				Xp	17

Xp = Primary Issues Addressed
Xs = Secondary Issues Addressed

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Figure 4-1. Issues & Experiments

Figure 4-2 summarizes the experiment classes, with regard to the purpose, definition and impact on issues. A discussion of each class and group of experiments is presented in the following paragraphs.

4.2.1 Class I - Voice Network Interface

This class of experiments is designed to address the problem of several existing voice networks interfacing as a single Defense Switched Network (DSN). By fully defining the electrical interfaces and signaling and supervision sequences between networks, intelligent decisions can be made about future upgrades of existing switches, or requirements for new switches. After determining which networks are of primary interest to the DSN (Autovon, FTS, etc.), and which interfaces on those switches are most significant, a series of experiments will be run that will define the inherent limitations of an interface to that network. This will lead to a definition of the required changes to achieve that interface. The test bed will not be concerned with jurisdictional or tariff issues, or restrictions, but will concern the technical possibilities if and when these impediments can be overcome. The basis of these experiments are the result of system studies such as the SETA program or other DCA studies.

The objective of these experiments is to verify interface possibilities between existing voice networks. Some specific goals are:

a. To provide the maximum possible information about interconnectivity between networks

b. To consider minimum use of operators and elimination of second dial tones in a network

c. To extend the features between networks by expanding the numbering plan; e.g., by using digit insertion, deletion or conversion

d. To implement automatic dialing with automatic use of features whenever possible. Precedence/Preemption, conferencing, call transfer, and call forwarding are examples of features that will be tested. As a result of these tests the operational capabilities between networks can be specified and decisions made concerning future upgrade of existing switches or requirements for new switches.

Class of Experiments	Purpose	Definition	Impact on Issues
1. Network Interface (voice subscribers)	To verify interface possibilities between voice subscribers in order to utilize all networks	Perform calls from each network to all other voice subscribers. Make direct calls plus calls involving premption, conference and other features	• Interoperability will be determined so interfaces can be specified • Transitional plan will have well defined options • Extension of features to other networks can be specified
2. Routing	To verify routing possibilities and strategies for circuit and packet switching. Recommend routing algorithms including feature limitations	Perform calls to test routing of tandem calls through various networks, and terminating on ones' own network, or another network. Determine restrictions and limitations of features as calls are routed through external networks. Test various packet routing techniques including dynamic routine over a floating trunk	• Survivability will be improved by the routing of calls around damaged trunks or networks • Interoperability will be determined for the tandem case • Cost factors can be limited by the use of other networks and/or spare trunks to alleviate the heavy load during busy hours

Figure 4-2. Summary of Experiments

Class of Experiments	Purpose	Definition	Impact on Issues
2. Routing (Cont.)			• Transitional plan can include routing options • Extension of features during tandem calls can be specified • Routing for combined voice and data can be specified
3. Combined Voice and Data	Determine the interface hardware and software control necessary to service simultaneously, both voice and data subscribers using present day equipment	Investigate the provision of voice and data communication using standard T1 transmission format	• Integrated switching concepts - new capabilities in the network and more novel forms of communication tion e.g. voice/graphics terminal. • Syst. Control/Net. Mgmt use of CCIS for controlling the provision of several services on one node
4. Common Channel Signaling (CCS)	Develop a common channel signaling scheme for use in the DSN	Experiment with on-net calls to verify CCS operation with DSN features (eg. MLPP)	• Recommend formats and data collection techniques for system control and network mgmt

Figure 4-2. Summary of Experiments (Cont.)

Class of Experiments	Purpose	Definition	Impact on Issues
4. Common Channel Signaling (CCS) (Cont.)		Experiment with off-net calls to ensure that compatability with CCIT No. 7 has has been preserved	• Specify efficient use of bandwidth to reduce transmission costs and increase network capability • Improve operational capability by specifying techniques for advance network features
5. Network Monitoring and System Management	To measure and evaluate the effectiveness of different, fault monitoring, testing and reporting techniques and network control methods for different types of networks	Compare distributed and centralized network control and monitoring. Examine different data transport schemes for effective need and impact. Examine centralized control recovery strategies	• Define requirements and reporting methods for monitoring the network • Present trade-off data for central versus distributed control and verify operational strategies for network resource allocation

Figure 4-2. Summary of Experiments (Cont.)

Class of Experiments	Purpose	Definition	Impact on Issues
6. Advance Voice Experiments	Determine efficient methods of voice transmission, i.e. ways to increase the number of conversations carried by a transmission link	Investigate the use of TASI and packet switching techniques for transmitting voice	• Cost-more efficient use of transmission facilities • Integ. Syst. Concepts - improves performance of data on integrated voice/data link; allows development of all-packet system • Syst. Control/Net. Mgmt. - CCS used to control TASI
7. Data Experiments	Evaluate complexity in providing different levels of protocol translation in a data network	Examine the aspects of central or distributed protocol conversion. How many levels of protocol can be effectively translated? What are the best candidates for network interconnections	• Combine the different data services into one network

Figure 4-2. Summary of Experiments (Cont.)

Class of Experiments	Purpose	Definition	Impact on Issues
8. Hybrid Experiments	Determine the hardware and software control for servicing both voice and data subscribers using hybrid switching techniques	Investigate use of TASI to facilitate variable allocation of bandwidth for data within a standard T1 frame structure Investigate the use of longer frame lengths and variable bandwidth channels for increasing transmission efficiency	Integrated switching concepts ● Recommended strategies for bandwidth allocation ● Recommended frame length ● Identify frame mgmt. techniques

Figure 4-2. Summary of Experiments (Cont.)

Experiments will be run from two points of view. The first assumes that the test bed is a given switch type, e.g., DDD, and is attempting to interface with other networks (see Figure 4-3a). The test bed subscriber will place calls to the network under test, attempting to use his own, or the test network's, features. The test bed will change or translate signals as required to be compatible with the other network. Results of a given test are the changes required for compatability and the services or capabilities achieved. This series of tests will be run for all likely candidates of network configurations. Results can be used to define proven interface for future switches.

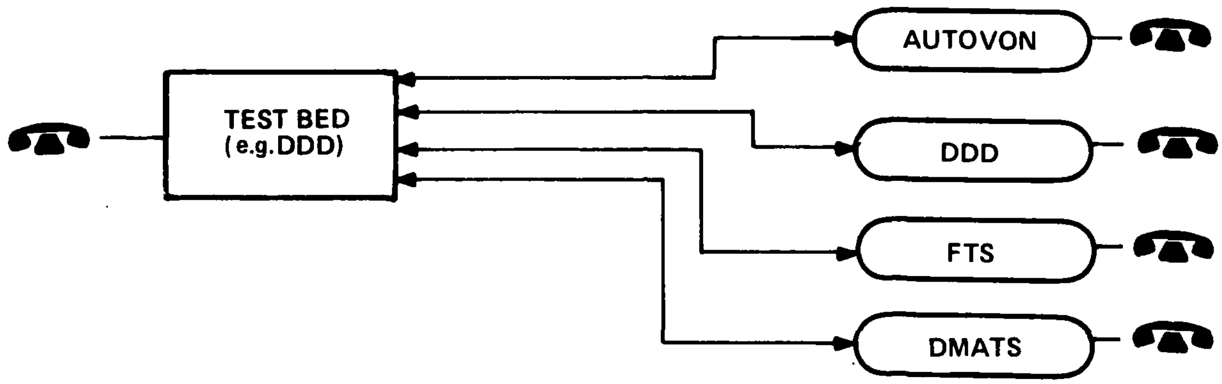
The second viewpoint is that of an applique, or black box addition, that allows interface to another network (Figure 4-3b). In these tests, calls will be placed from one network to the other through the "applique test bed" the applique will perform all numbering-plan conversion and signaling translation as required. The intent is to allow existing switches to operate with the newer switches in an upgraded fashion and to define the translation algorithms that would be implemented in any new switches. In this way, suggested changes can be made in existing equipment to further enhance applique operation.

Actual tests to be performed could evolve as the test bed is developed; however, the following is a partial list of tests that can be performed:

<u>Group 1</u> - Test Voice Network Interfaces

Test A: The test bed or prime caller will place a direct routed call to the external network using each of the chosen interfaces. For example, a DDD interface would allow a call to be placed from the test bed, or through the "applique test bed", to all chosen AUTOVON interfaces.

Test B: The prime subscriber will preempt terminals in order to access trunks or subscribers. During this test any preemption limitations, or alternatives to pre-emption, such as converting precedence to call-waiting or barge-in feature in the DDD, will be specified.



(a) Test Bed Assumes Given Switch Type

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(b) Test Bed Acts as Interface to Other Network

Figure 4-3. Voice Network Interface

Test C: The prime subscriber will initiate various types of conference calls such as preset, progressive, and meet-me conferencing. These calls will be extended to all possible networks and interfaces.

Test D: The prime subscriber will exercise all available features into all test networks in order to determine feature differences among networks. For example, what will happen when a DDD subscriber exercises Camp-On Busy into the AUTOVON network, which does not have the camp-on feature? Therefore, this test will define the limitation or new specifications needed for the interoperability of these interfaces.

4.2.2 Class II - Routing

This class of experiments is designed to address routing strategies and problems between networks for voice and data traffic. Recommended routing algorithms will be tested, feature capabilities verified, transmission quality recorded, and packet routing techniques compared. Routing techniques for alternate routing, least cost routing, survival routing and routing through satellites will also be explored.

The goals in determining routing strategies are as follows:

a. To increase the probability of call completion by being able to route calls around damaged or busy trunks and networks,

b. To verify operational capabilities thus allowing for intelligent decisions to be made concerning future switches and their routing capabilities,

c. To cut costs by verifying possible use of alternate networks or flexible trunks instead of having to size each network for the peak or busy-hour loading,

d. To implement flexible routing that can be used for defining transition plans in order to move from analogue switches and multiple networks to digital switches and integrated networks.

The first group of experiments addresses the problems of alternate routing when a call in one network is destined for a subscriber in a different network. Class I experiments assumed that

the call exited to the destination as soon as possible. The techniques to be investigated here would have the call transit as far as possible within a newtork before exiting. The first group also addresses alternate routing restrictions due to satellite delays. If information about satellite routing cannot be passed between networks, restricted use of satellite trunks in originating networks would have to be employed.

The second group of experiments addresses the problems of having a call originating and terminating in the same network using external networks for tandeming. The problems of transparency and multi-shuttling will be investigated.

The third group of experiments tests the strategies and effectiveness of various routing methods within existing packet networks. The network survivability and overhead efficiency are affected by PAR and PVC techniques. Routing table update strategies and mixing of formats will be investigated.

The fourth group of experiments will develop a flexible circuit-switches 56 Kb/s T1 channel to be used for offloading traffic during congestion or during periods of delay. The following points will be considered:

a. Central or distributed network management of the flexible trunk. For example, the DIN network can determine what the topology looks like and which calls should use the flexible trunk.

b. Possible utilization of non-leased public network or AUTOVON trunks for calls can be considered.

c. The efficiency of a non-dedicated trunking facility will be examined (e.g., resulting Bit Error Rate, queue backup problems, call set-up delays, redistribution of traffic).

Below is a list of routing experiments to be performed on the test bed. Groups 1 and 2 (see Figure 4-4) involve routing of voice calls and Groups 3 and 4 (see Figure 4-5) involve routing of data calls.

Group 1 - Test routing possibilities for switches to route calls to their own networks through various trunks (satellite and terrestrial) or to other destination networks by delayed exit.

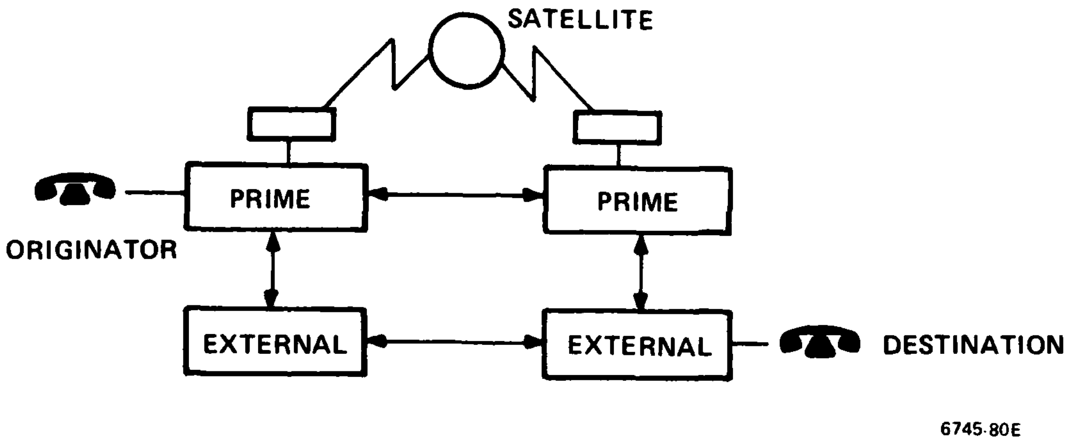

Figure 4-4. Use of the Test Bed for Voice Routing Experiments

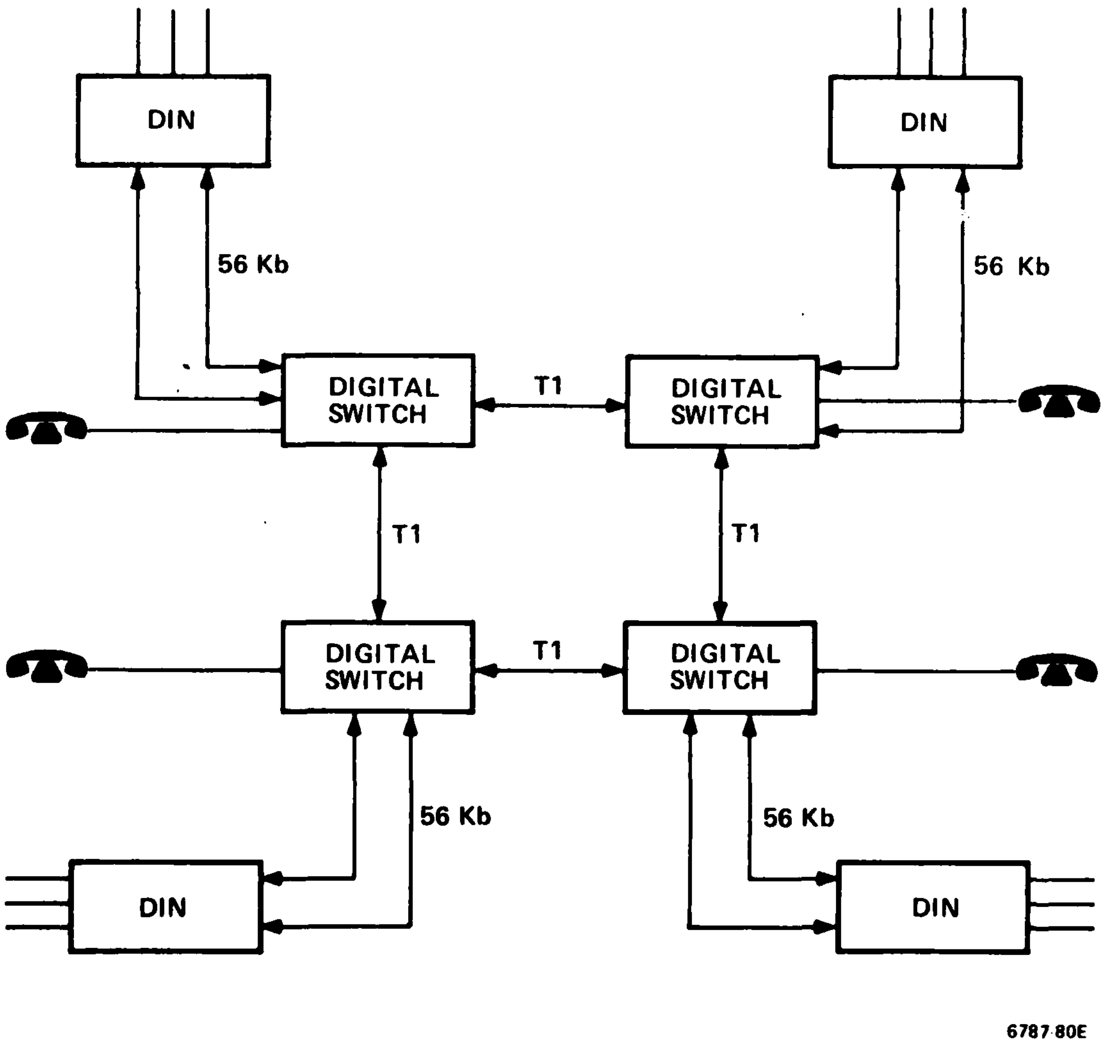

Figure 4-5. Use of the Test Bed for Data Routing Experiments

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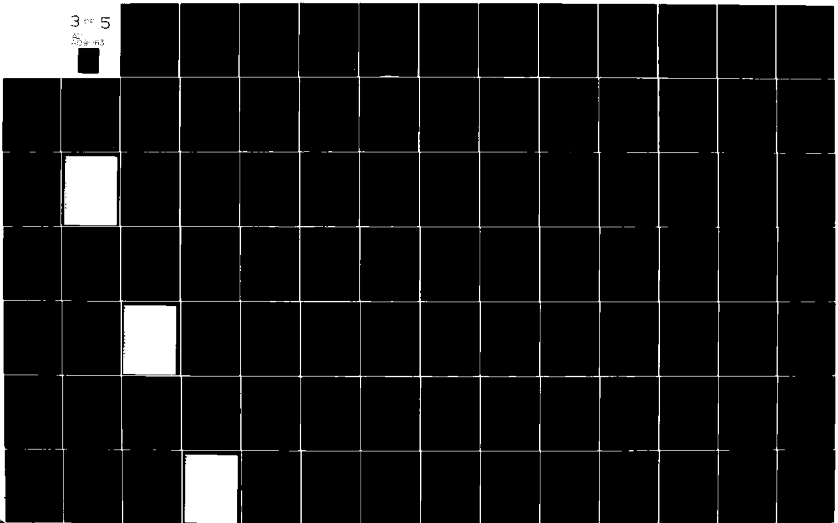

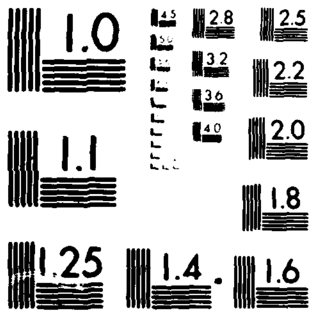

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Test A: Place voice calls within the calling network (PRIME) or to an external network by routing the call through a trunk in the calling (PRIME) network first, in order to get as close as possible to the destination before exiting to the external network. The call is then routed to the subscriber in the external network.

Test B: Place all calls that are routed over either satellite or terrestrial trunks and measure the effects and differences (e.g. - delay).

Test C: Place calls with different blocking conditions for the various routes (e.g. - equipment busy and blocking condition in prime and external networks) and mesaure the relative performance. This includes tests involving originating office control and spill-forward office control.

Test D: Place tandem encrypted calls to verify special routing algorithms for such security calls.

<u>Group 2</u> - Test routing strategies for using other networks for tandeming calls.

Test A: Place tandem voice calls from one subscriber to another subscriber through an external network using all possible features. The two subscribers may be in the same network or in different networks. These tests will be used to define restrictions necessary for placing tandem calls through external networks.

Test B: Place tandem voice calls in order to study least cost routing by:

1. Testing least cost routing when only cost is considered
2. Testing least cost routing when features must be maintained.

Test C: Test emergency routing algorithms by placing tandem voice calls with feature implementation as a secondary goal and call completion as the primary goal.

Group 3 - Test packet switching routing techniques

Test A: perform test calls that involve different methods for the routing of packets. Evaluate the following types of packet switching routing techniques:

1. Packet Adaptive Routing (PAR) technique where each packet is routed separately.

2. Packet Virtual Circuit (PVC) routing technique where the first "trailblazer" packet is routed deterministically or adaptively and all subsequent packets are routed along this same path.

3. Mixed routing where both PAR and PVC techniques can be used. For example, interactive data may be routed using the PAR technique and bulk data may be routed using the PVC technique.

Test B: Perform tests that measure efficiency of the error detection and retransmission algorithms of different packet switching routing techniques.

Group 4 - Test dynamic packet switching routing to take advantage of a flexible trunk availability.

Test A: Perform experiments using an available flexible (floating) trunk and develop a dynamic packet switching routing scheme to take advantage of this trunk. This type of movable trunk could exist by virtue of connection to DAMA satellite or the backbone circuit network. A typical test would require that the load to a particular destination be increased or that an existing trunk be damaged, requiring that the system allocate an additional trunk to that destination.

4.2.3 Class III - Combined Voice and Data Experiments

This class of experiments investigates methods of combining voice and data communication using standard transmission facilities and switches, with new interfaces and software modifications provided as required. One key issue addressed here is Integrated System Concepts, but with emphasis on the nearer term as opposed to the more exotic integrated voice/data techniques, e.g., packet switching of voice and data. There is also an impact on the issue of System Control/Network Management as well. The objective is to obtain some of the advantages of integrating voice and data services within the bounds of existing equipment and networks. Some specific goals include:

a. Integrated voice and data sharing of transmission facilities
b. Higher speed data service from point to point, e.g., 56 kbps facsimile or bulk data transfer
c. Improvement of the interaction between circuit and packet switching networks
d. The ability to quasi-associate two services in a switch, i.e., voice and data (interactive, graphics, video, FAX) with one subscriber.
e. Common network monitoring and system control to do the above (See Class V).

Two groups of experiments are proposed for addressing the issues of Integrated System Concepts and System Control/Network Management relative to the DSN. The first group of experiments will investigate ways of combining voice and data on standard T1 trunks (see Figure 4-6). Voice at 64 kbps PCM will coexist with data on the same trunk, with data occupying one or more 56 kbps channels of the T1 link (out of a possible 24 channels). Data calls are treated as if they were voice calls, i.e., circuit switch protocol is used for setting up and breaking down all calls.

The second group of experiments (see Figure 4-7) partitions users into two classes: those requiring circuit switched service (e.g., voice/FAX) and those requiring packet switched service (e.g., interactive data). Packet switched service is regarded as simply an add-on feature to the network of existing digital circuit switches. Requests for transmission bandwidth (one or more T1 channels) by

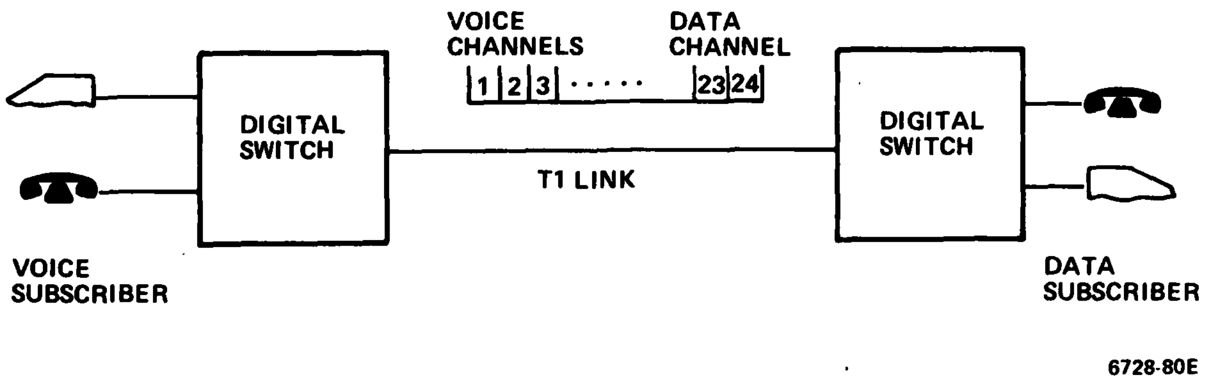


Figure 4-6. Combining Voice & Data on Standard T1 Link

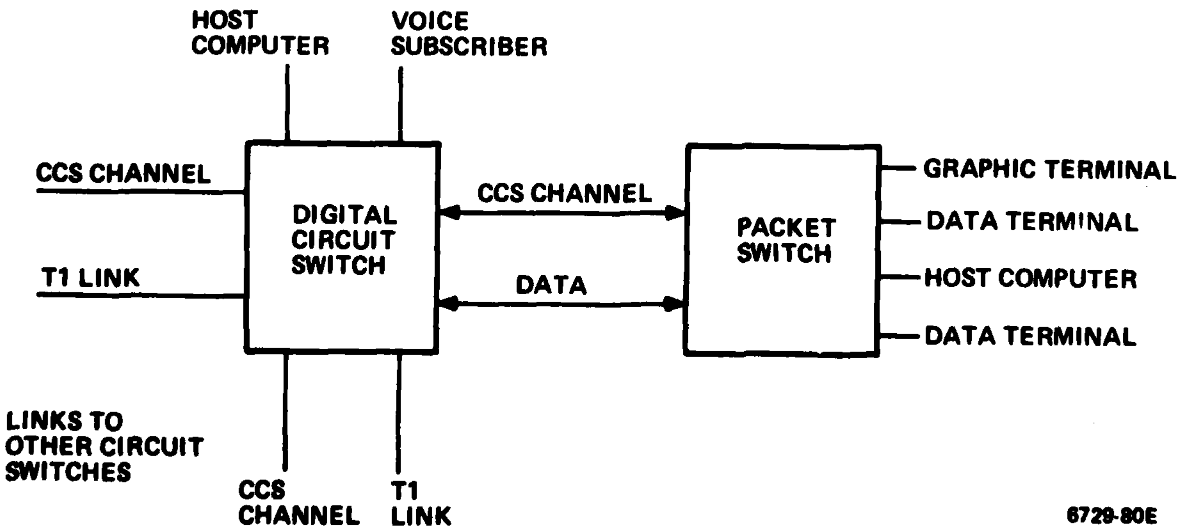


Figure 4-7. Combined Voice and Data Switching

packet switched data subscribers are made through the circuit switch protocol. However, once capacity has been assigned, information transmission is carried out using packet switching techniques. A key element is the development of the CCS protocol and message format for controlling the packet-to-circuit-switch network interface. Development of this capability will help ease the transition to an integrated traffic environment while making maximum use of existing facilities.

Group 1 - Test methods for carrying voice and data on standard T1 trunks.

Test A: Using a standard Bell T1 trunk (D2/D3 format), assign on demand 56 kbps channels to individual data calls.

Test B: Establish protocol and traveling classmarks necessary to maintain an all-digital path between data subscribers. Identify any routing restrictions, network control and A/D conversion problems as a result of providing this service.

Test C: Establish terminal interface to digital subscriber protocol for interaction between data subscribers and the digital switch.

Test D: Reconcile the different voice/data precedence schemes.

Group 2 - Test combined voice and data switching using circuit switching and packet switching service.

Test A: Scenario in which packet switch requests more bandwidth to a given destination. Additional capacity is established through the network of circuit switches using common channel signaling.

Test B: Measurements are taken of the duration of call setup of a data subscriber at the packet switch through the network using the circuit switch setup procedure. The capability of the procedure to adequately handle various data types, e.g., interactive bulk, at a low level of delay service, is determined.

Test C: Scenario in which interswitch trunk failure occurs. Circuit switch informs packet switch of failure, re-establishes new connection and informs packet switch of such.

Test D: Scenario in which packet switch trunk module fails. Circuit switch is interrogated; then the same T1 trunk channel is allocated to a spare packet switch module.

Test E: Establish a voice/graphics call between subscribers via telephone. Demonstrates interaction of voice/data features and provides an example of the new services obtained by combining different traffic types in the same network.

Test F: Evaluate various ways of communicating data measurement and control information to a centralized monitoring and control center. For example, should data be collected at the packet switch, put in packet format and transmitted directly to a control facility, or should it be sent through the circuit switch via the CCS channel?

Test G: Evaluate interaction of bandwidth requirements for voice and data, i.e., whether the circuit or packet switched subscriber gets priority for bandwidth.

Test H: Ship bulk data from host computer by circuit or packet switching means. Measure the impact on efficiency of bulk data transfer under each type of switching technique. The efficiency will vary with different line speeds.

Test I: Measure the effects of different Bit Error Rates in the circuit network on overall efficiency due to retransmissions, errors, etc.

4.2.4 Class IV - Common Channel Signaling Experiments

This class of experiments is aimed at developing specifications for a common channel signaling scheme for use in the DSN. Common

channel signaling is recognized as a technique having a number of advantages over current in-band signaling techniques. These advantages include:

a. Cost reduction through use of common signaling equipment as well as faster call setup signaling which helps reduce call holding time.

b. Reliability through error detection techniques and retransmission.

c. Greater information carrying capacity than conventional in-band signaling techniques, lending to its suitability for use as a Syscon or network management data link.

d. Flexibility in expanding messages to permit implementation of new network features.

Two groups of experiments are planned for addressing issues in the DSN. Both groups involve investigation of and experimentation with techniques for enhancing the capabilities of the common channel signaling scheme while at the same time preserving compatibility with the existing CCITT Signaling System No. 7.

The first group of experiments is aimed at developing or modifying existing or proposed CCITT No. 7 formats to include: travel classmarks for route selection and to control connections for voice and data calls; Messages used for system control and/or network monitoring. This group of experiments will provide data on which to base intelligent decisions regarding the implementation, in the DSN, of common channel signaling compatible with CCITT No. 7.

The second group of experiments is aimed at developing criteria for new features, and the required CCS enhancements to implement these features (e.g., Network Call Forwarding, Distributed Conferencing, etc.). These experiments will provide techniques leading to enhanced operational capability as well as more efficient use of network transmission bandwidth, thereby lowering operating costs.

The CCS experiments should, in the aggregate, cover the following aspects of CCS:

a. The message formats and use. This is known as the Telephone User part (TUP), and would involve the inclusion of traveling classmarks, precedence indicators, new messages (e.g., PREEMPT) and the like. [1]

b. The signaling techniques. This is known as the Message Transfer Part (MTP). Various techniques to be considered are:

   1. Separate signaling links

   2. Alternate methods of signaling over T-carrier such as bit-stealing from individual channels vs. use of an entire individual channel.

   3. In the far term, the use of a constant-bandwidth dedicated channel vs. an on-demand variable-bandwidth channel vs. packetized CCS.

<u>Group 1</u> - The first group of experiments on formats will include the following tests:

Test A: Test bed subscribers will place on-net calls involving common channel signaling between switches over separate signaling links. Tests will involve both routine and precedence voice calls as well as data calls.

Test B: Test bed subcribers will place off-net calls involving common channel signaling between the test bed switch and the external network. These tests will involve only routine calls.

Test C: Tests will be run to evaluate use of CCS for originating office routing control vs. spill forward control.

Test D: Tests will be run to evaluate use of quasi-associated CCS for data calls, satellite call setup, system control, offnet tandem calls, etc.

Test E: Tests will be run to evaluate different formats of data collection from switches.

Test F: Tests will be run to evaluate different recovery strategies for various categories of failures via use of CCS.

Group 2 - The second group of experiments for CCS feature control will include the following tests:

Test A: Tests will be run to evaluate centralized vs. distributed data base for different features such as:

1. mobile subscribers
2. call forwarding
3. multi-homes subscribers
4. multi-network subscribers

Test B: Tests to verify conference setup and control when distributed through the network such that minimum trunking is required (each group of conferees homes on their own switch and then the conference bridges linked by trunk).

Test C: Tests to evaluate different conferencing algorithms (who links or homes to whom for conferencing, spill forward conferencing).

Test D: Tests of Broadcast (multi-addressed) data connections for FAX, etc.

4.2.5 Class V - Network Control and System Management

This group of experiments addresses the problems and possibilities associated with network monitoring and system control. There are several areas of interest included in this group: AMA, distributed versus local system control, automatic tech control (built into the switch), and various data reporting techniques. Experiments are applicable to both the near and far term and are directed at techniques for the existing and next generation circuit and packet switches, and future experimental switches. Running these experiments will enable DCEC to be able to evaluate different approaches to tech control and network management, making maximum use of automatic techniques. Thus requirements for the next generation equipment or even about-to-be purchased equipment can be better defined. The objectives and plans for total system management of the

DSN can also be better defined through the use of experimentally proven information. Some specific goals of system control and network monitoring are:

a. To provide lower operation and maintenance cost,

b. To increase reliability and survivability,

c. To provide accurate and complete data on the present network for future growth and planning,

d. To provide full network control for Hybrid networks.

Operation and maintenance costs are principally affected by tech control features. Fast and accurate testing of switching equipment such as receivers and terminal circuits, and general background diagnostics, are obvious enhancements for maintenance reduction and are often a standard part of any new switch. More advanced aspects of this service would be the automatic routine testing of lines and trunks for quality and operation. Other features would include the reporting of outages and maintenance requests to both local and remote central locations. Some questions exist with regard to the methods for transfer of the reported information to the Network Monitoring Center, particularly those using CCS (Group IV).

Other experimental tasks will evaluate the relative merits of central control versus distributed local control of failures and network status changes. For the advanced integrated voice/data systems, experiments will be extended to the areas of network resource allocation based on various parameters, such as precedence, outages, and congestion. These likewise will be evaluated for the relative merits of local versus distributed control for both reliability and effectiveness. The ultimate system would be a completely self testing and self controlling network that would report failures by optimizing routing and general resource allocation. Additionally, it would provide complete system performance characteristics under various network conditions so that weaknesses could be identified and improvements defined.

Three groups of tests will be run using various test bed configurations.

The first group of experiments will deal with existing switches. Experiments will consist mainly of developing techniques to provide failure detection and reporting in a manner that is transparent to the existing equipment. The data from this type of testing, plus any data that is available from the switch itself, will then be transferred to either the central or local reporting facility, depending on the experiment.

The second group of experiments will deal with the more advanced circuit switches to be specified for the DSN.

They should have greater inherent diagnostic power. They should also be able to deliberately select the route over specific trunks on a scheduled-determined or traffic-determined basis to provide a much higher fault detection and isolation capability. Experiments in this group will evaluate the trade-off between having faults analyzed locally with summary reports sent to a remote station and having all faults sent to a remote station for analysis. Similar types of experiments could also be run for the packet switching networks. The objective would be to determine the parameters that are common to the two types of networks and those that are unique. This would enhance message and statistic formatting by making maximum use of a common system control protocol. The evolution of the network toward integrated voice and data switching would thus be more easily accommodated using this standard protocol.

The third group of experiments are directed at the problems of integrated voice/data network control. Primary interest is resource allocation during periods of contention.

A partial list of experiments to be performed is given below:

<u>Group 1</u> - Provide external test equipment at each switch in the network and connect these to a central control facility as shown in Figure 4-8.

Test A: Have each test box operate independently as a special terminal on the system. Each unit will place calls, keep track of time of day, call attempts, delay to

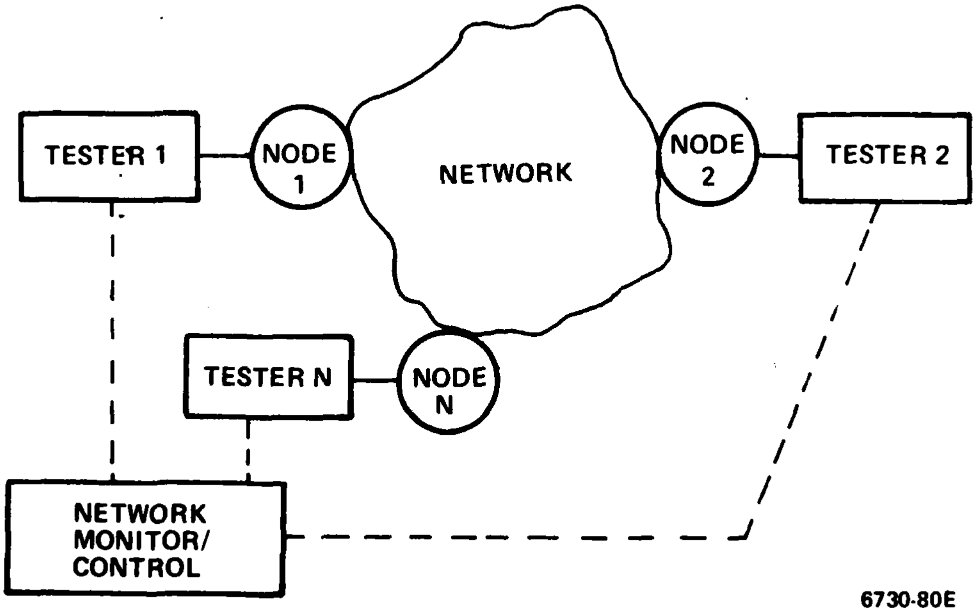


Figure 4-8. Direct Interconnect of the NMC and Distributed Switches

answer, transmission quality, equipment busy, terminal busy and other parameters that may be measurable.

Test B: With the same test equipment but under a central control, run the same types of experiments. Coordination and instantaneous fault isolation and diagnostic methods can then be evaluated.

<u>Group 2</u> - Investigate centralized and distributed techniques for system control and network monitoring in which the switches and network are included as active parts of the test equipment.

Test A: Tests consist of calls placed through specific routes with special routing codes and numbers or traveling classmark features used to enhance testing.

Test B: With a similar network capability as in A, calls will be placed without specifying particular routes.

Test C: For the above tests, the information obtained will be transmitted between individual switches or test units and the central NMC (Netowrk Monitoring Center). The format of this information will either be in summary form with the local equipment doing some pre-processing, or will be a more raw data form of transmission in which the NMC is required to do the processing. A trade-off between centralized versus distributed processing will be made with regard to transmission efficiency and effectiveness.

Test D: Comparative evaluation of a Central control and its decision algorithms, and distributed, interconnected and independent forms of netcontrol will be tested under various failure and overload recovery conditions.

Test E: An alternative to the configuration of Figure 4-8 for direct interconnect of the NMC and distributed switches in a limited network access is shown in Figure 4-9. This network transmission method will be evaluated for efficiency and control delay.

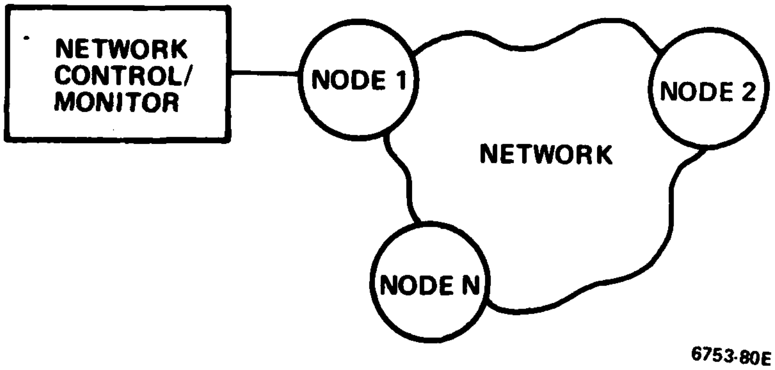


Figure 4-9. Limited Access of the NMC to Network Switches

Test F: The test bed will be used to evaluate the effect of including AMA and overhead types of traffic on the NMC channels. The effects and overhead of automatic, requested and event driven reporting will be evaluated under different network load conditions.

<u>Group 3</u> - Investigate Hybrid (Circuit/Packet) Network Control

Test A: Hybrid networks may have a special use for Central system control and monitoring. A network such as the one in Figure 4-10 could make good use of a central facility to reconcile the demands of the data and voice networks. The experiments would evaluate different control algorithms that would allow more or less bandwidth for data trunking.

4.2.6 Class VI - Advanced Voice Experiments

This class of experiments will investigate advanced techniques for transmitting voice waveforms more efficiently. These include Time Assigned Speech Interpolation (TASI) and the packet switching of voice. Issues impacted by these experiments include Cost, Integrated System Concepts, and System Control/Network Managment. Both TASI and packetized voice allow a more efficient use of transmission facilities and hence, have the potential for cost reduction. An interesting point is whether or not this improvement in voice efficiency can lead to improved data performance in terms of delay and throughput. The introduction of packetized voice allows the development of an all-packet system, in which voice and data are both packet switched, although they may use different routing algorithms or error control strategies. A new protocol must be developed to meet the more stringent delay requirements for voice packets that takes advantage of the robustness of the speech waveform for error control.

Two groups of experiments are planned. The first group will concern the development of TASI capabilities for T1 trunks (Figure 4-11). Various forms and formats of common channel signaling will be used to identify the talkspurts from a particular conversation that

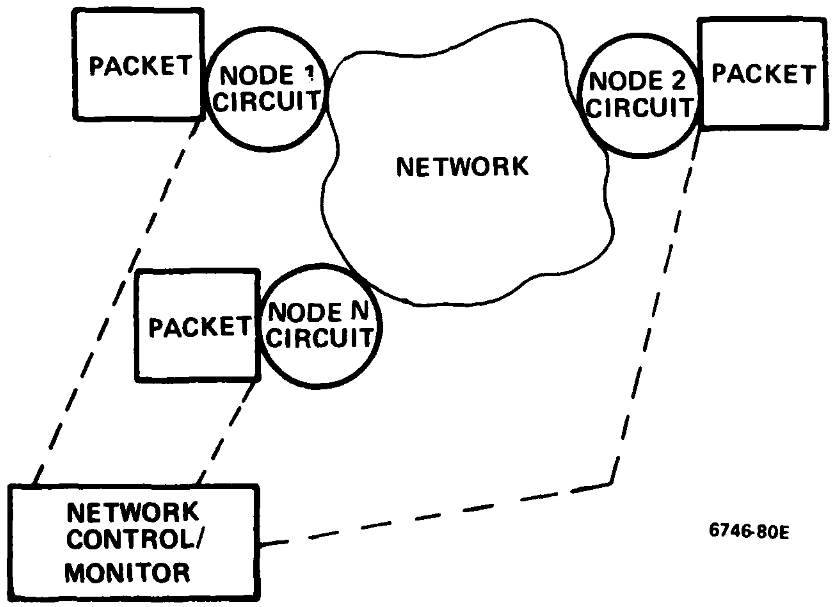


Figure 4-10. Interconnection of the NMC in a Hybrid Network

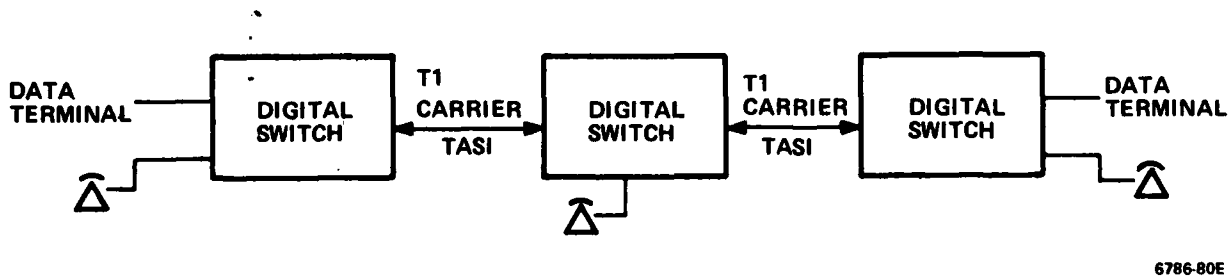


Figure 4-11. TASI Capability for T1 Link

are assigned to a given channel at any instant in time. Only fixed bandwidth voice calls, i.e., 64 Kb/s PCM, will be used. Success in these experiments would allow an increase in the number of voice conversations carried by a given transmission link.

The second group of experiments examines the use of packet switching for transmitting digitized voice. The efficiency of a packet switched network in processing low bit rate subscribers is a key measurement, since the ratio of the control header to actual voice information is much larger than it is for high bit rate voice at a fixed packetized rate. Delay is a potentially significant problem in networks with many users, especially for voice packet systems. Although the overhead efficiency can be improved by using fixed length packets, there would be added delays in the call. The psychological effect of delay on talkers is well known from experince with satellite links and it will be important to determine the trade-offs between efficiency and delay. During experiments it will be necessary to measure not only end-to-end delay, but also buffer utilization, nodal processing; and other contributors to delay, to determine where development must be applied to reduce excessive delays in the transmission chain. Another factor in voice communications is synchronous voice continuity. In a packet system, dispersion of the packets along the transmission path, caused by the interpolation of voice packets for other conversations, appears to be the prime contributor affecting voice continuity. Measurement of packet dispersion and ultimate correlation with subjective listening tests will be necessary to determine the effect of dispersions on listener perception and thus the bound, if any, that this places on the size and distribution of the user population. Related to the factor of voice continuity is voice quality as a function of bit errors, and lost, blocked, delayed, or misdelivered segments or packets. Just as for continuity testing, ultimate correlation of subjective listening tests with measurements of erroneous behavior will be a key to determining the effectiveness of each concept under these conditions.

Group 1 - Experiments to test TASI on a T1 trunk.

Test A" Measure the effects of TASI, using different TASI update rates, on T-carrier (24 channels and larger) link-by-link transmission.

Test B: Measure the effects of TASI on tandem calls, particularly multiple hops and combined trunks.

Test C: Measure the effect and means of restricting TASI for data calls and security calls, e.g., using classmarks. For example, when data is on a trunk, does the performance of TASI get worse because of a lack of channels?

Test D: Investigate the best method for controlling TASI, e.g., new T1 format with common channel signaling, using a control channel for more than one T1 carrier.

Test E: Measure the clipping effects of TASI for different voice digitization schemes (e.g., CVSD and LPC vs. PCM).

Group 2 - Experiments to test packetized voice:

Test A: Examine different packetizing and delivery algorithms for effectivness on unencrypted voice.

Test B: Evaluate different packet protocol issues for voice, e.g., PAR vs. PVC routing algorithms, giving voice packets precedence over data packets.

Test C: Measure delay and delay dispersion for different classes of packetized voice and data.

Test D: Evaluate speech quality and voice continuity.

Test E: Evaluate throughput and overhead efficiency.

Test F: Evaluate different flow control techniques for packetized voice and other packet data.

Test G: Evaluate the effects of packetizing on encrypted speech and data.

4.2.7 Class VII - Data Experiments

The data conversion experiments are intended to show the feasibility of combining several data services and protocols onto a single network. There are several levels where this can be accomplished in a layered protocol system. Level I (electrical)is probably the easiest in that it is accomplished at the line level and is well defined. A subtler aspect of Level I protocol is line speed variations which are more easily handled in a packet network than in a circuit-switced network.

Level II protocol (link level) is more difficult but still manageable. One difficulty that could present problems is byte-oriented to bit-oriented protocol conversion. Interface data transfer between switch and subscriber and the network can be handled but the information data would be more difficult due to the non-conforming format.

Level III (network level) protocol is the most difficult to translate since basic message information and forms of error checking can vary.

The goal of this feature in a network is to provide universal connectivity between terminals using different protocols. This would greatly increase the use of existing equipment and expand the interconnection of different services.

The tests defined here are mainly verifications of paper studies. These are directed at determining the feasibility and complexity of providing different levels of protocol conversion. Tests are further directed at measuring the effects that translating protocol has on overall performance. Principal areas of suitability would appear to be in the packet switching domain, but some circuit switch applications could arise.

The two principal ways of attacking the protocol conversion problem are (a) to translate at either the originating or terminating terminal to the other terminal's protocol, or (b) to translate at both ends of the communication link. The latter method appears more reasonable since a more universal translation can take place.

Questions of bit-oriented or byte-oriented protocol translation are a key problem. Tests will also be concerned with the possibilities of bit stuffing and destuffing.

Group 1 - Data Experiments

Test A: Interoperate two terminals of different speeds and electrical interfaces (level 1) that are using the same level II & III protocols.

Test B: Interoperate two data subscribers that are using different Level I & II protocols but are using the same Level III protocol. There are several possibilities for experimentation including Bisync to X.25 protocol conversion.

Test C: These tests are geared at performance evaluation. The effects of translation and the effect different protocols have on over-all performance can be measured. The network may not want to do link-by-link error detection if higher-level protocols are sufficient. The overall efficiency of data transfer can be measured under different network bit error conditions.

Test D: A series of tests that will assume that the host computer is able to use several level III protocols. The network would have the responsibility of translating Level I and II protocol and, as part of the Level II, would inform the terminating equipment which Level III was in effect. This would be particularly useful in dial-up service between remote terminals and host computers.

Test E: Interoperate two data terminals in different networks or in different subnetworks of the same network. Investigate the problems associated with internetwork connection, including the areas of addressing, routing, flow control, error control and terminal access.

Test F: Investigate the provision of secure communication among data terminals located in the same network or in different networks.

4.2.8 Class VIII - Hybrid Experiments

This class of experiments will investigate the use of hybrid switching techniques for integrating voice and data. One key issue addressed here is Integrated System Concepts. The issues of cost and transition strategy are also addressed. The overall objective is to investigate ways of automatically allocating bandwidth between voice and data in order to maximize transmission efficiency on the trunks within acceptable bounds of blocking of circuit-switched traffic and delay of packet-switched traffic.

Two groups of experiments are planned for addressing the issues of Integrated System Concepts and Cost for the DSN. The first group of experiments also addresses the issue of transition strategy because it can be accomplished using existing 64 Kb/s voice equipment. This group of experiments will investigate the characteristics of a hybrid switching scheme incorporating a T1 subframe structure within a longer (e.g., 10 msec.) master frame in which the T1 subframe map will remain constant for the entire master frame interval. Circuit switched voice channels will be transmitted in the early part of the subframe, with the remainder of the sub-frame available for transmission of packet switched data. The boundary between the two classes of traffic will be adjusted every master frame, thus maximizing the availability of unused circuit switching bandwidth for transmission of packet switched data. The further enhancement of combining TASI and TADI will also be investigated. The frame structure is shown in Figure 4-12. (See Class VI for TASI results).

The second group of experiments will investigate the effects of variable frame length and variable bandwidth contiguous channels on speech quality, voice and data delay, and transmission efficiency. The circuit-switched voice channels will be transmitted in the early part of the frame and the packet-switched data in the remainder of the frame. The boundary between the two regions will be variable from

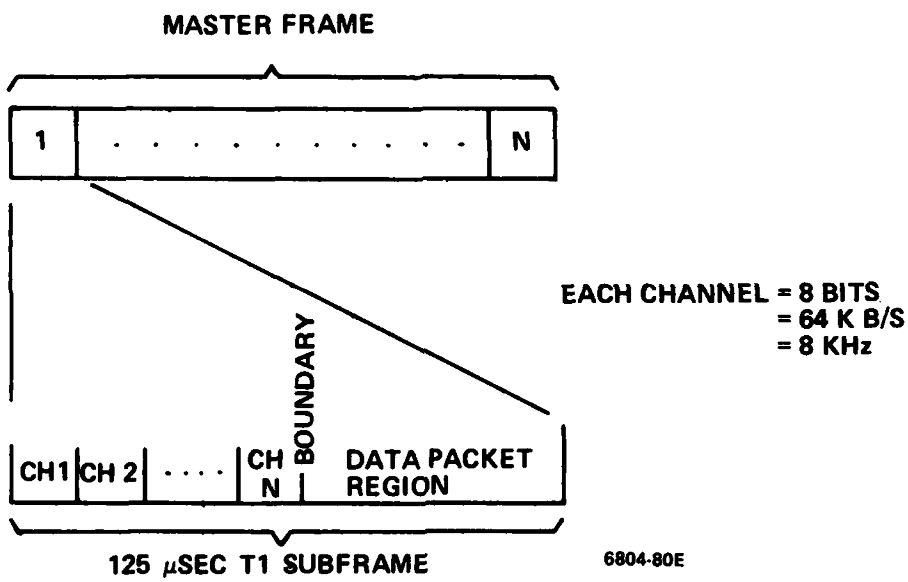


Figure 4-12. Frame Structure - First Group

frame to frame. The frame structure is shown in Figure 4-13. The primary difference between the frame structure of the two groups of experiments is that the master frame in the first case consists of an integral number of T1 sub-frames, and the samples of a given channel are therefore distributed throughout the frame. The master frame structure in the second case carries all of the samples of a given channel contiguously in the frame. It is obvious that the second scheme has a potential for greater delay dispersion than the first.

<u>Group 1</u> - Experiments to investigate characteristics of fixed bandwidth and variable boundary T1 will include the following tests:

Test A: Verify that operation with fixed bandwidth, variable boundary, and TASI is possible.

Test B: Evaluate different strategies for bandwidth allocation (flow control, precedence between voice and data, non-TASI circuit bandwidth for encrypted voice, etc.).

Test C: Evaluate and measure efficiency with different TASI ratios (how many calls per 24 channels?) with different data offered and different TASI rates.

Test D: Measure effects on delay for packet data.

<u>Group 2</u> - Experiments to investigate variable frame/variable boundary switching will include the following tests:

Test A: Test the effects of different frame sizes on delay and dispersion for voice and speech quality. Consider multiple tandem hops. In this testing, only constant bandwidth channels are to be used, and TASI is not to be applied.

Test B: Apply TASI to the constant bandwidth voice calls, utilizing a TASI map which is transmitted each frame at the beginning of the master frame. This map will indicate the active/inactive status of each of the

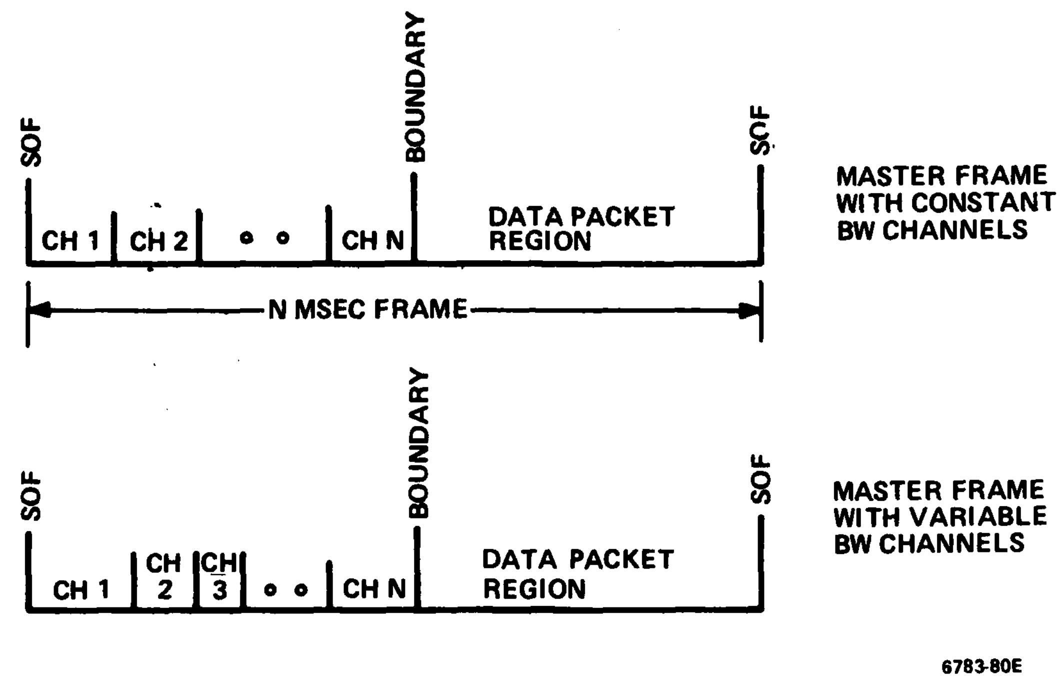


Figure 4-13. Frame Structure - Second Group

voice calls being serviced. Thus, the circuit-switched/packet-switched boundary will vary dynamically every frame as a result of the cumulative TASI actively across all of the circuit-switched region.

Test C: Evaluate variable bandwidth calls (vocoder, etc.) in the circuit region with no TASI. Compare with the packetized voice calls of experiment Group VI.

Test D: Measure and evaluate quality and efficiency of fixed bandwidth with TASI versus variable bandwidth with no TASI.

Test E: Evaluate performance with variable versus fixed boundary.

Test F: Develop and evaluate precedence and preemption strategy in variable bandwidth schemes.

REFERENCES FOR SECTION 4

(1) CCITT Signaling System No. 7, Draft Recommendation Q.721, November 9, 1979.

SECTION V

TEST BED ATTRIBUTES AND REQUIREMENTS

5.1 INTRODUCTION

This chapter discusses the minimum test bed attributes and requirements necessary for conducting each of the Experiment Groups described in the preceding chapter.

The experiment groups as described in Section IV fall into eight classes comprising a total of seventeen groups. Figure 4-1 presented, in matrix form, the experiment classes and groups vs the issues addressed by each group, as well as the recommended order in which the experiment groups are to be performed. The relative order is based both on the immediacy of the issues being addressed, and on the interdependencies of the experiment groups themselves.

Figure 5-1 shows the relative order and interdependencies of the various experiment groups. It should be noted that the interdependencies are based not on incremental test bed configurations, but rather on results of earlier experiments being required for subsequent experiments. Figure 5-1 represents the case in which the experiments are performed strictly in the same sequence as their relative order. The figure as drawn, however, shows the potential for a high degree of concurrency. Figure 5-2 illustrates a case at the other end of the spectrum in which each experiment group is performed as soon as possible, regardless of the desired relative order. Figures 5-1 and 5-2 thus bound the range of possible sequencing. The necessary tradeoffs will be made and discussed in subsequent chapters.

The following paragraphs of this chapter describe the minimum requirements for each experiment group on a stand-alone (or bottoms-up) basis. In order to minimize redundancy, a requirements list is presented in Table 5-1 containing various capabilities required in more than one experiment group. This list is referred to in the discussion of each experiment group.

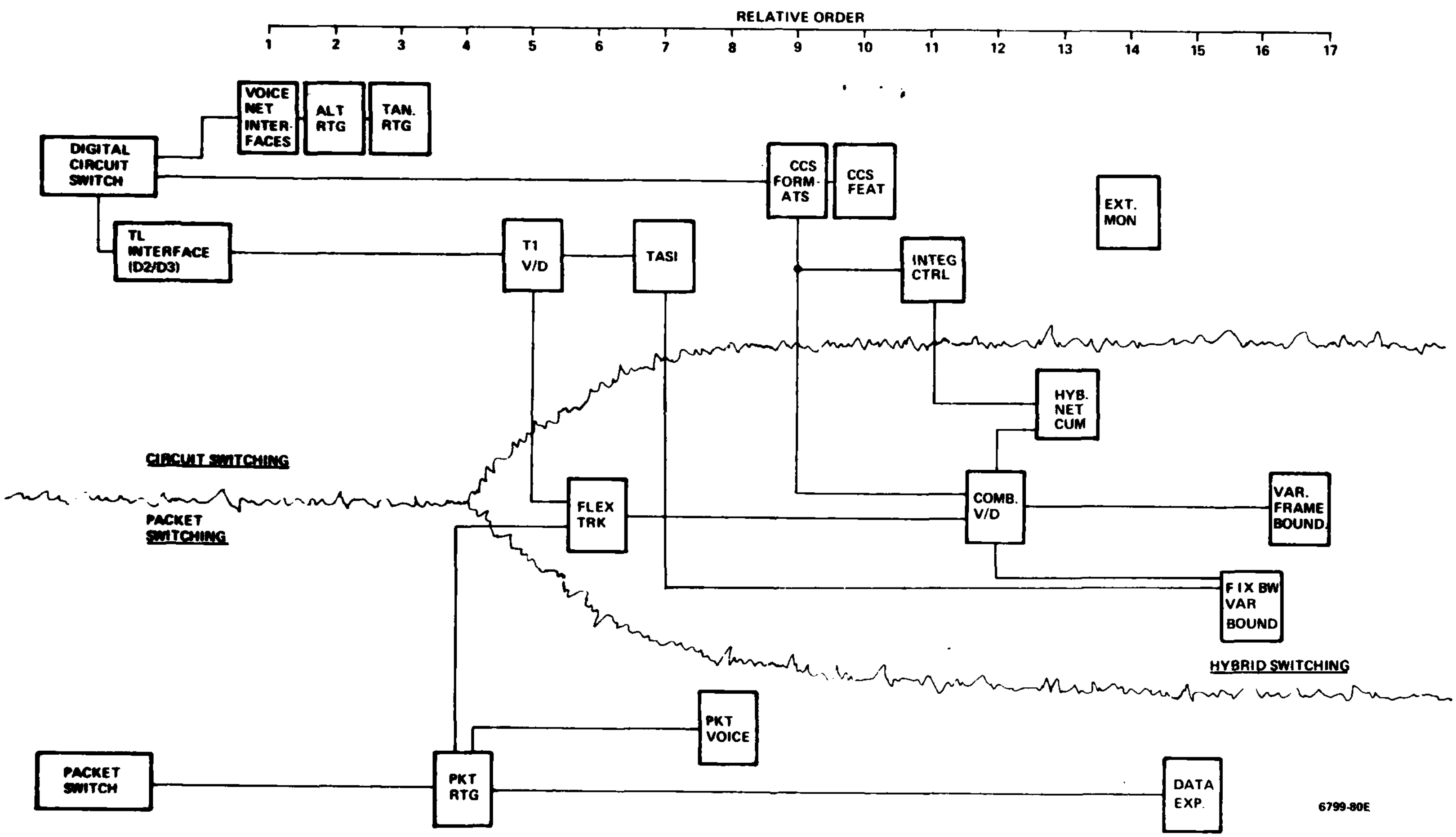


Figure 5-1. Interdependencies & Relative Order of Experiments

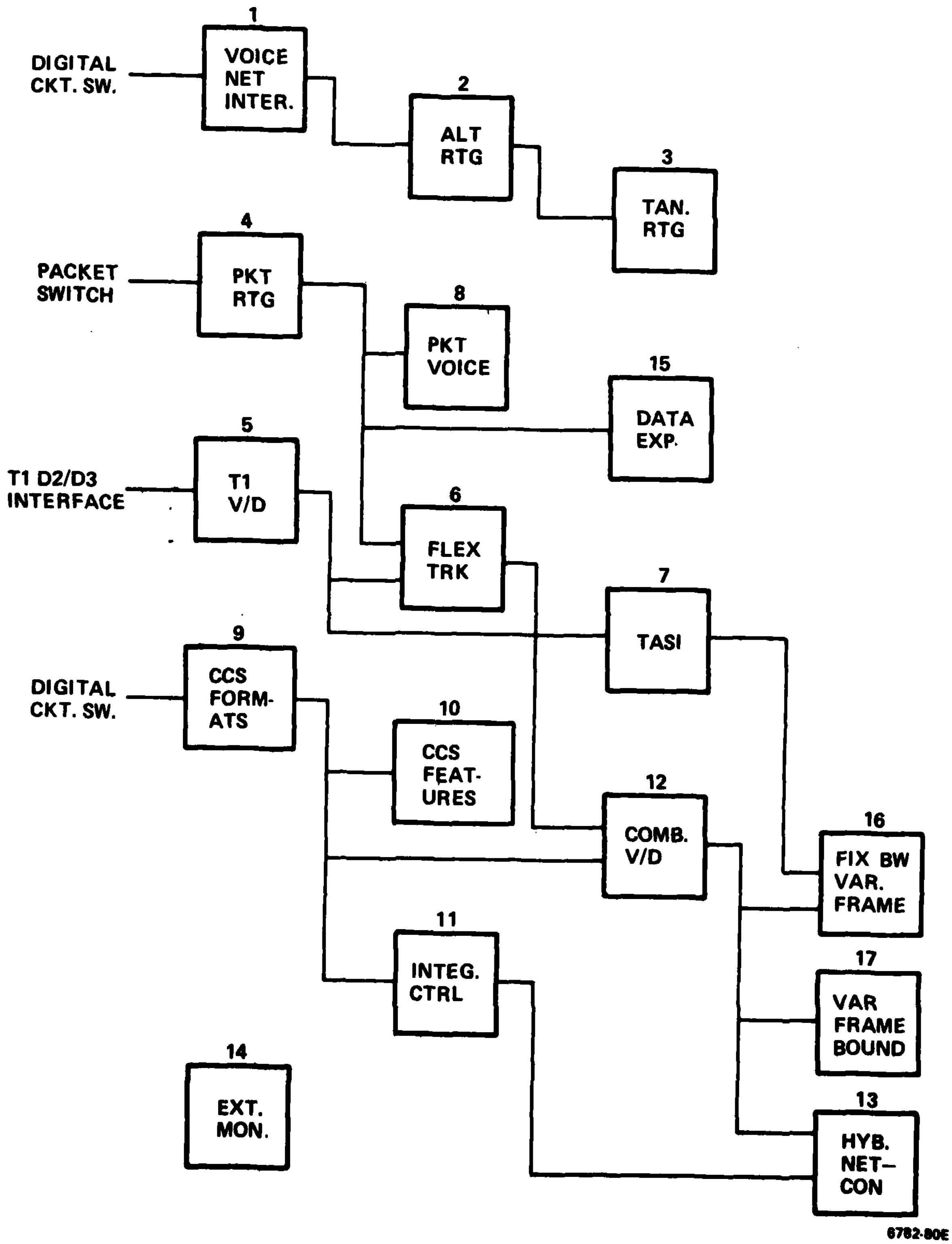


Figure 5-2. Experiment Sequencing with Maximum Concurrency

TABLE 5-1. REQUIREMENTS LIST

A. 4-Wire Digital Circuit Switching Capability:

Minimum of 20 lines/trunks
2-wire/4-wire subscribers

a) 3 DTMF with 4 x 4 keypad
b) 3 rotary dial

Stored program control
Minimum 7-digit numbering plan
Deterministic routing (Alt routing not req'd)
Handle minimum of 10 simultaneous calls
Data base modifiable locally (by user)

B. Feature Processing Package

Five-party conferencing
Operator
Call Forwarding
Camp-on-busy

C. Multilevel precedence and Preemption Capability

D. Data Subscribers (RS-232 interface)

E. Means of Measuring BER

F. T1 D2/D3 Interface

G. Voice Network Interfaces (HW & SW)

Autovon trunk
Autovon PBX
E&M trunk with DP signaling
E&M trunk with DTMF signaling

H. Satellite Interface (HW & SW)

I. Alternate Routing Capability (Based on Classmarks, Number Conversion, Etc.)

J. Tandem Routing Capability

K. Packet Switching Capability

Multiple 56 KBS trunks
x.25 interface (HW & SW)
Packet type data subscribers

TABLE 5-1. REQUIREMENTS LIST (Cont'd)

L. Different Packet Routing Techniqes

- PAR
- PVC
- Combination

M. Packet Traffic Loading Capability

N. Packet Traffic Measurement Capabilities

- Delays
- Retransmissions
- Volume

O. Forced Error Capability

P. Packet Switch/Circuit Switch Communication Link

Q. Common Channel Signaling (CCITT #7)

R. Centralized Monitor/Control Facility with Data Links to Each Switch

S. Automated Transmission Test Equipment

T. Broadcast (Multi-Addressing) Data Capability

U. Circuit Traffic Loading Capability

V. Speech Activity Detector

W. TASI T1 Trunk (HW & SW)

X. Speech Encoding Interfaces (eg. LPC, CVSD, Etc.)

Y. Fixed BW, Variable Boundary T1 Trunk

Z. ILRAN Trunk

- Variable BW
- Variable Boundary
- Longer Frame (eg. 10 msec)

AA. Voice Packet Assembly/Disassembly (PAD) Unit

BB. Various Data Subscriber Interfaces

- HDLC
- BISYNC
- ADCCP
- SDLC

Finally, the last paragraph presents the characteristics of a full-up test bed able to perform all of the experiment groups previously described.

5.2 INDIVIDUAL EXPERIMENT GROUP REQUIREMENTS

5.2.1 Voice Network Interfaces - Class I, Group 1

Minimum Requirements:

Item	A	B	C	F	G	J
Quantity	2	1	1	4	1	1

Experimental Dependency:

None

Discussion

Experiments will consist of placing calls from subscribers and trunks in one network to another, carrying forward as many features as possible. Features required are only necessary to the extent that they apply to inter-net calls. By this token, routine capabilities are required only to the extent that fixed directory numbers are routed to certain trunks and that trunk signaling incoming from one trunk can go out on another trunk. The switch must be able to vary timing and sequencing at the trunk interfaces. It must also be capable of assuming the characteristics of the various networks for emulation. Routing and translation shall consist of digit insertion, number substitution and the performance of various signalling sequences based on originating and destination terminal types and feature utilization.

Possible exceptions to these listed requirements are the need for a T1 carrier, depending on the defined interfaces of the study, and the absolute necessity of a digital matrix. A digital matrix is desirable, but a simple analog matrix would suffice for the voice grade network interfaces being studied since these are analog.

5.2.2 Alternate Routing - Class II, Group 1

Minimum Requirements:

Item	A	B	C	E	F	G	H	I	J
Quantity	3	1	1	1	6	1	1	1	1

Experimental Dependency:

5.2.1

Discussion

The test bed will simulate a three node network with two nodes in an originating network and a third node in a terminating network. Experiments make use of the interface requirements defined in Section 5.2.1 and expand the routing and call sequence algorithms to make use of multiple networks for transmission. The switches shall be capable of using originating office and spill-forward control. Alternate routes will be chosen on a deterministic basis but selections must be based on originating call features, terminating network and special digit routing codes. Satellite interfaces are treated as another trunk with special characteristics. Switches shall be capable of special digit insertion/modification to perform traveling classmark functions.

The tests in this group include measurement capability to determine error and delay of calls on specific trunks and routes. These are for information only and are not part of dynamic routing algorithms.

The same restrictions and possible exceptions on the use of T1 trunks and digital switching apply here as for Section 5.2.1.

5.2.3 Tandem Routing - Class II, Group 2

Minimum Requirements:

Item	A	B	C	E	F	G	H	I	J
Quantity	3	1	1	1	6	1	1	1	1

Experimental Dependency:

5.2.1, 5.2.2

Discussion

The test bed must be capable of simulating three nodes with two nodes in the home network and a third node acting as an external network. The tests will consist of originating and terminating calls in the home network tandemed through external networks. Switches shall be capable of routing based on features, termination, call type, etc. Switches shall also be capable of new signaling sequences and similar routing capabilities to those described in 5.2.2. An example of a tandem call placed from an Autovon to Autovon subscriber through the DDD network is as follows:

The originating Autovon Switch places a call to the destination Autovon Switch by dialing a special subscriber number through the DDD. When the DDD completes the call to the terminating Autovon Switch, the terminating switch recognizes the special number and transmits a proceed to send signal to the originating switch. The real call signaling information is transferred and the call is completed. Answer supervision etc. is returned to the DDD as required. The effect is one of layered signaling.

Exceptions to the requirements and call data gathering capability are identical to those in 5.2.2.

5.2.4 Packet Routing - Class II, Group 3

Minimum Requirements:

Item	K	L	M	N	O
Quantity	3	1	1	1	1

Experimental Dependency:

None

Discussion

The packet network shall be capable of various routing strategies. Trunk interfaces shall be capable of implementing different error control mechanisms such as CRC. There shall also be a capability to change the strategy of error retransmission depending on traffic type. The capability to handle different sized headers and different traffic types is also a requirement. (See MIXD Concept, Section III).

The test bed shall have the capability to measure traffic parameters and various characteristics associated with packet data. These include delay, retransmissions, packets per trunk, number and type of packets and total bits transmitted. The test bed shall have the capability to load different trunks and switches with traffic in order to test various routing algorithms. There shall be a capability to force selected errors at a specified rate to test error checking schemes and network control.

5.2.5 T1 Voice/Data - Class III, Group 1

Minimum Requirements:

Item	A	C	D	E	F	I
Quantity	2	1	2	1	1	1

Experimental Dependency:

None

Discussion

Switches shall have the capability to interface directly to a data subscriber using a digital format. Precedence tables shall be alterable to vary the relative order of data and voice precedence. Switches shall also be capable of special signaling or number conversion to handle traveling classmarks and other designators indicating the presence of data calls (rather than voice calls). Data calls can assume a transmission rate of 56 Kb/s with any speed conversion handled by the user multiplexer.

5.2.6 Flexible Trunk - Class II, Group 4

Minimum Requirements:

Item	A	D	F	I	J	K	L	M	N
Quantity	3	1	1	1	1	3	1	1	1

Experimental Dependency:

5.2.4, 5.2.5

Discussion

Packet and circuit switches will be connected through a modified digital subscriber interface. Modifications are required to either the packet switch or circuit switch digital interfaces, or both, to allow signaling and supervision exchange in order that the packet switch looks like a standard data subscriber to the circuit switch. The packet switch shall be capable of modifying its routing algorithms to analyze traffic patterns in order to determine traffic flow. In the event that a new trunk to a particular destination would relieve traffic congestion, the packet switch shall be capable of establishing this new trunk by simply placing a "Call". It would then re-route traffic as appropriate. The test bed shall measure packet traffic statistics during the transition. Delay of setup and relief of traffic congestion shall also be determined.

5.2.7 TASI - Class VI, Group 1

Minimum Requirements:

Item	A	F	J	U	V	W	X
Quantity	3	1	1	1	1	1	1

Experimental Dependency:

None

Discussion

The test bed shall be capable of being configured as a three-node circuit switch network. Switches shall have voice activity detectors on each subset and trunk. Trunks shall be capable of performing TASI using several control mechanisms and different TASI rates. Switches shall be able to adjust maximum TASI assignments and restrict different channels (such as data) from having TASI applied. Switches shall also have terminals (1 per switch required) that use other voice encoding algorithms besides 64 Kbps, such as CVSD or LPC, so that TASI effects on these types of coding can be studied.

5.2.8 Packetized Voice - Class VI, Group 2

Minimum Requirements:

Item	K	L	M	N	O	V	X	AA
Quantity	3	1	1	1	1	1	1	1

Experimental Dependency:

2.5.4

Discussion

The test bed shall be capable of being configured as a three-node packet switch network. The test bed shall incorporate voice packetizing/depacketizing units in the network. Switches shall be capable of routing calls for various numbers of tandem hops. Voice packets shall be transmitted according to the different transport mechanisms (PAR, PVC, CRC, NON-CRC, ETC.) of 2.5.4. The packet network shall be capable of loading trunks with various mixes of voice/data traffic. Main results required are correlation between the measured data parameters of 2.5.4 and the subjective effect of these parameters.

5.2.9 CCS Formats - Class IV, Group 1

Minimum Requirements:

Item	A	D	G	H	I	J	Q
Quantity	3	1	1	1	1	1	1

Experimental Dependency:

None

Discussion

The test bed shall have the equivalent of three nodes interconnected by common channel signaling links. Maximum use of CCITT#7 shall be made, but the signaling scheme must be adaptable to be compatible with other network signaling schemes. Trunk access to other networks is required but the signaling for these connections shall be made by CCS. Test bed switches shall be capable of quasi-associated, originating office and spill forward control. Network monitoring and control messages shall be accomodated by the CCS formats. Routing shall be variable based on network condition determined by CCS messages, originating and terminating location, features used, previous satellite hops and other special conditions.

5.2.10 CCS and Features - Class IV, Group 2

Minimum Requirements:

Item	A	B	C	D	I	J	Q	R	T
Quantity	3	1	1	3	1	1	1	1	1

Experimental Dependency:

5.2.9

Discussion

The test bed shall be capable of being configured as a three node circuit switch network. The common signaling (CCS) plan developed in 5.2.9 shall be employed with the message formats extended to control remote conference set up, broadcast data and other features as they are developed in the test bed. The centralized facility shall be connected by CCS data links and will use the common CCS formats to communicate. Switches shall be capable of sending broadcasting data to multiple receivers for FAX and other similar data traffic. It is desirable to accomplish this without use of a conference bridge since a return path is generally not necessary. Requests for retransmission can be done by CCS.

5.2.11 Integrated Control - Class V, Group 2

Minimum Requirements:

Item	A	D	I	Q	R	S	U
Quantity	3	1	1	1	1	1	1

Experimental Dependency:

5.2.9

Discussion

The test bed shall be capable of being configured as a three-node circuit switch network. Switches in this configuration shall be capable of formatting and reporting data to a central facility by quasi-associated CCS or by direct CCS links. The test bed shall be capable of measuring the bandwidth requirements for the CCS channels under the two network configurations: directly connected and quasi-associated connected. The test bed shall also be capable of loading the test network and monitoring the effectiveness of recovery strategies by distributed switch control and centralized control. The test bed shall measure the effects on network operation when overhead maintenance traffic such as AMA (real or simulated) is added to the CCS channels. Measurement capability shall include trunk utilization, call set up delay, etc. under the different control conditions. Under either configuration (distributed or centralized control) the switches shall be capable of line load control, adjustment of routing tables and call restriction.

5.2.12 Combined Voice/Data - Class III, Group 2

Minimum Requirements:

Item	A	C	F	I	J	K	L	M	P	V
Quantity	3	1	1	1	1	3	1	1	1	1

Experimental Dependency:

5.2.6, 5.2.7

Discussion

A CCS channel shall be established between the circuit and packet switch at each node. An extended format of CCITT#7 shall be used and shall be compatible with previous CCS formats. A similar functional capabilty to that of 5.2.9 is envisioned. The additional intercommunication between the two switches should result in improved efficiency and adaptability. The test bed shall be capable of loading the packet switch with traffic to force a request for extra bandwidth. The test bed shall also be capable of loading the circuit switch trunks with dynamic call requests in order to force contention for bandwidth between the circuit and packet switched calls. Data collection to determine call delay, packet delay and traffic loading with relation to trunk loading is required. The bandwidth contention algorithms shall be capable of varying voice/data priorities.

Information for transmission efficiency comparisons shall be provided. The capability to force difference bit error rates in data transmission for both packet and circuit switched data is required to allow comparative evaluation of efficiencies at different data rates. Efficiency measurements of CCS vs. packet transport of control data is also required.

5.2.13 Hybrid Network Control - Class V, Group 3

Minimum Requirements:

Item	A	D	I	K	L	M	N	P	Q	R	S	U
Quantity	3	1	1	3	1	1	1	1	1	1	1	1

Experimental Dependency:

5.2.11, 5.2.12

Discussion

The test bed shall use the capabilities of 5.2.12 and add tne capability to control a hybrid network. Packet switch control, statistics reporting, and testing coordination are added to the control portion of the central control facility. The central control facility and the hybrid nodes shall be capable of transferring all data on CCS channels (packet switch informs circuit of transmission) or via packet as determined in 5.2.11.

The test bed shall have the capability to measure the net efficiency of controlling overall network transmission facility. Arbitration of bandwidth assignment by central and ditributed control shall be possible.

5.2.14 External Monitoring - Class V, Group 1

Minimum Requirements:

Item	A	D	H	J	R	S
Quantity	3	2	1	1	1	2

Experimental Dependency:

None

Discussion

The test bed shall be capable of being configured as a simple circuit switch network. It shall be capable of placing calls along different routes. The test equipment shall be connected to a central control point and shall use the signaling formats discussed in 5.2.9 to exchange control and data information. Calls shall be placed under software control at various test boxes. The test boxes shall recognize the call sequence tones of the network, measure call setup time, measure line quality and establish different error conditions for data in the network. The test box shall also be able to define where and when (time) error conditions are encountered in the network for a particular call.

5.2.15 Data Experiments - Class VII, Group 1

Minimum Requirements:

Item	D	F	H	K	L	BB
Quantity	1	1	1	2	1	1

Experimental Dependency:

5.2.4

Discussion

The test bed shall be capable of interfacing several level 1, 2, and 3 protocols in the packet network. Switches shall translate between the network and language protocols. Data collection capability for bit error rates, lost packets, traffic frequency, etc. shall be available to evaluate secure traffic protection.

5.2.16 Fixed Bandwidth/Variable Boundary - Class VIII, Group 1

Minimum Requirements:

Item	A	C	H	K	L	M	N	P	Q	R	U	V	W	Y
Quantity	2	1	1	2	1	1	1	1	1	1	1	1	1	1

Experimental Dependency:

5.2.7, 5.2.12

Discussion

The test bed shall be capable of being configured as a three-node network. The nodes of the test bed shall be connected by a new type of "Modified T1" trunk. D2/D3 format is not required but 8 kHz (125 µsec) multiplexing for each channel is required. The test bed shall have the capability of compacting the active voice channels toward the beginning of the frame, and using the remaining bandwidth as a variable rate packet trunk. The boundary between the circuit-switched and packet-switched regions shall vary by 64 KBS increments. TASI shall be applied to voice classmarked lines. The test bed shall be capable of loading the trunk for both circuit-switched and packet-switched calls. The test bed shall also be capable of measuring the delays, loading, etc. of both circuit and packet traffic to evaluate the interaction and control algorithms.

5.2.17 Variable Frame/Variable Boundary - Class VIII, Group 3

Minimum Requirements:

Item	A	C	H	K	L	M	N	P	Q	R	U	Z
Quantity	2	1	1	1	1	1	1	1	1	1	1	1

Experimental Dependency:

5.2.12

Discussion

The test bed shall be capable of being configured as a three-node network. The test bed nodes shall be interconnected by the new "Modified T1" trunk. TASI capability is required for fixed but not for variable bandwidth, circuit switch calls. Any unassigned circuit region bandwidth is automatically assigned to packet switched traffic. The trunk multiplexing algorithm shall treat circuit switched calls as contiguous data. The trunk shall have flexibility to vary the master frame rate. Measurement capability to determine the data transfer efficiency, call delay, set-up time, bit error rate and other trunk control characteristics is also required.

5.3 FULL-UP TEST BED ATTRIBUTES AND REQUIREMENTS

The complete full-up test bed necessary to perform all the experiments described thus far must satisfy the requirements identified in Table 5-1 and further broken down into hardware and software requirements in Table 5-2. This test bed shall include circuit switching and packet switching capabilities as well as advanced integrated voice and data capabilities. Figure 5-3 is a diagram that shows the test bed with a minimum of three nodes. All the required interfaces are shown for one of the nodes.

The mix of traffic in the full-up test bed shall include both real traffic and simulated traffic. The real traffic will include 2 and 4-wire voice subscribers that are local to the test bed and will include network interfaces for AUTOVON, DDD, FTS, and DMATS. Also included will be a wide range of data terminals and host computers using HDLC, Bi-Synch, X.25, ADCCP and SDLC protocol, as well as other packet switching protocols. The test bed will implement features that impact interswitch trunking such as conferencing, multi-addressing, multilevel precedence and preemption, and call forwarding.

A major function of the test bed shall be to implement packet switching and advanced integrated voice and data switching as described in previous experiment descriptions.

The test bed shall simulate traffic in order to produce sufficient traffic to saturate intersite trunks. This traffic emulation shall be designated to generate calls at specified rates with specific characteristics. Synchronous and asynchronous data traffic will be simulated and call set-up and tear-down will be performed for simulated circuit and packet switched calls. The test bed shall allow simulated voice and data calls to occur simultaneously with real traffic.

TABLE 5-2. HARDWARE AND SOFTWARE REQUIREMENTS

EXPERIMENT GROUP	HARDWARE REQUIREMENTS	SOFTWARE REQUIREMENTS
Baseline Digital Circuit Switch	2W/4W Capability 1 Node (Switch) 20 Lines/Trks (10 sim. calls) Interfaces: 3 DTMF Subscribers 3 Rotary Dial Subscribers 2 T1-D2/D3 Signaling Links with standard Signaling Digital Matrix	Basic PBX Capabilities Easily Programmable Options Stored Program Control Routing - Basic Translation Basic Commercial Interfaces
Class I, Group I Voice Network Interfaces 1.	Autovon Interfaces DDD Interfaces FTS, DMATS, etc. 1 5-party Conference 1 Operator	Software for Interfaces Feature Processing Conferencing Operators MLPP Call Forwarding Camp-On-Busy Others required by Study Numbering Plan/Translation for Features
Class II, Group I Alternate Routing 2.	2 Satellite Interfaces Total of 3 Nodes (2 additional)	Call Setup Timing Capabiity Alternate and Special Routing Capability Single Call Attempts
Class II, Group 2 Tandem Routing 3.		Least Cost Routing Tandem Routing
Class II, Group 3 Packet Switch Base and Packet Routing 4.	3-Nodes 6 56 KBS trks (2/node) 3 Interactive Data Terminals (Programmable rate) x.25 Interface	PAR Routing PVC Routing Mixed PAR/PVC Routing Retransmission Choices x.25 Interface Measure Call Delays, setup, etc. Routing Table Update Traffic Load Capability

TABLE 5-2. HARDWARE AND SOFTWARE REQUIREMENTS (CONT.)

EXPERIMENT GROUP		HARDWARE REQUIREMENTS	SOFTWARE REQUIREMENTS
Class III, Group I	5.	Data Subscriber Interfaces	Special Data Classmarks Error Measurement Ability MLPP for Data
Class II, Group 4 Flexible Trunk	6.	6 Packet/Circuit Switch Interfaces (DIN) (Packet Switch acts as Data Subscriber to Circuit Switch)	New protocol for DIN Routing for Flexible Packet Trunk Measure efficiency and usage of Spare Trunk Distribution of Network Loading with and without space capability
Class VI, Group I TASI	7.	New T1 Trunk for TASI Silence detector and Subscriber TASI Map Handler 1 LPC, CVSD, PCM	TASI Map processing External subjective measurements of TASI performance Variable Bandwidth
Class VI, Group 2 Packetized Voice	8.	Voice packetizing hardware	Extensive delay measurements
Class IV, Group I CCS Formats	9.	CCS link for circuit switches	MLPP & various features in CCS Data Control Quasi-associated routing Recovery strategy CCITT #7 (Modified)
Class IV, Group 2 CCS and Features	10.	More Subscribers Conference Bridge in each switch	Software for distributed and advanced features by CCS NMC for central database experiments Multi-addresses for FAX
Class V, Group 2 Integrated Control	11.	Automatic Test Equipment AMA and other statistical traffic (real or simulated)	AMA Expand self test diagnostic

TABLE 5-2. HARDWARE AND SOFTWARE REQUIREMENTS (CONT.)

EXPERIMENT GROUP		HARDWARE REQUIREMENTS	SOFTWARE REQUIREMENTS
Class III, Group 2 Combined Voice and Data	12.	CCS channel between circuit and packet switches Graphics Interface Common network control Traffic emulation (if external)	Emulator Software Performance measurements for bulk and interactive traffic Recovery strategy
Class V, Group 3 Hybrid Net Control	13.		New Software for NMC
Class V, Group 1 External Monitoring	14.	2 External test equipment/s for each node (could be spare node)	
Class VII, Group 1 Data Experiments	15.	Modify Data Interfaces To add new protocol chips	
Class VIII, Group 1 Fixed Bandwidth Variable Boundary	16.	New T1 format like TASI but puts packet data in unused area	New T1 Format Frame size change Measure delay and effiency for Packet and network efficiency
Class VIII, Group 2 Variable Frame Variable Boundary	17.	Modify T1 Mux Algorithm (new T1 trunks) Variable frame sizes	Variable frame sizes Constant bandwidth w/TASI capability Measurement ability Variable bandwidth w/o TASI

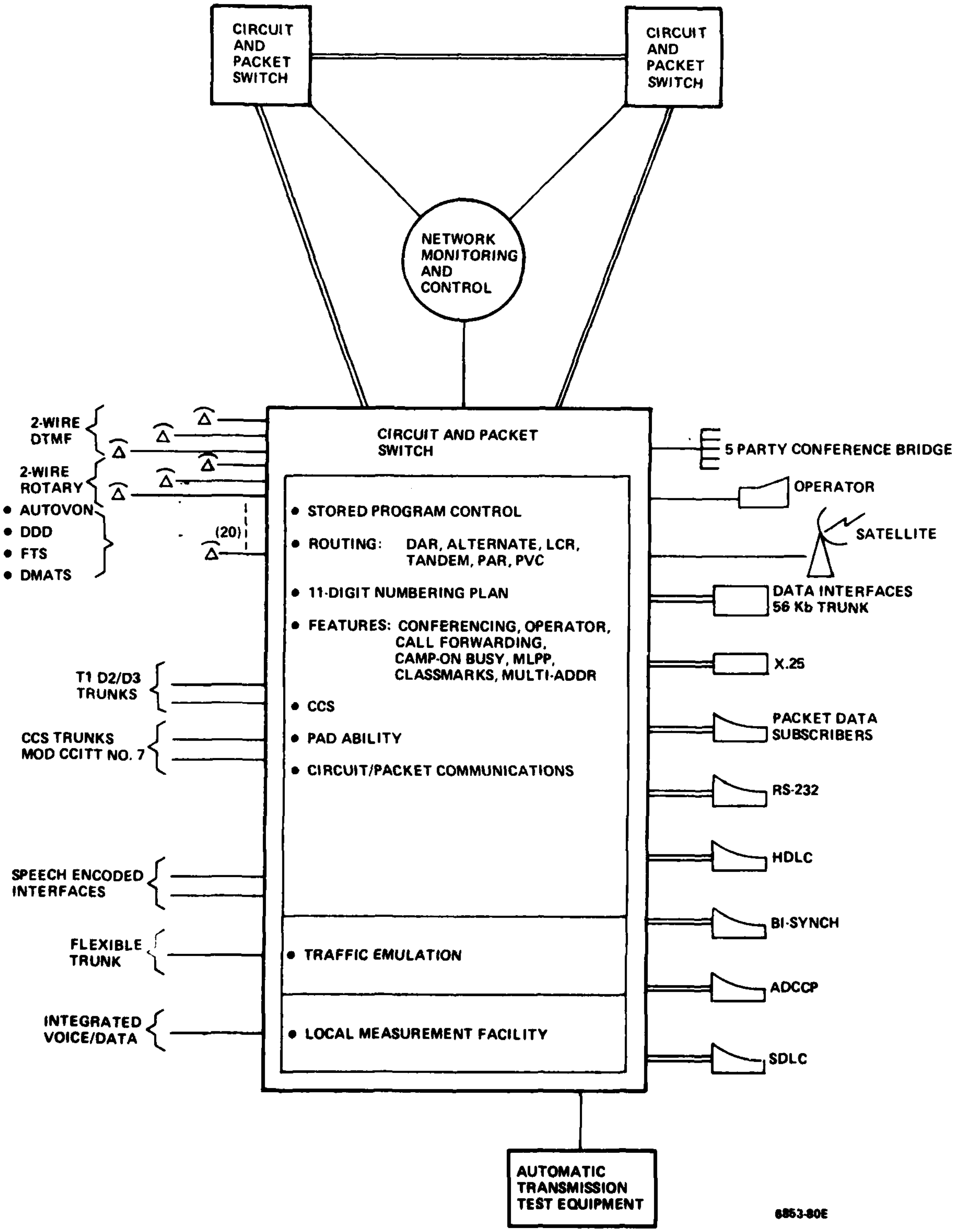


Figure 5-3. Full-Up Test Bed

Performance data for both real and emulated traffic shall be capable of being sent to a central monitoring and control center, depending on the specific experiment. This central facility shall be capable of gathering the data and presenting it in a statistical summary. In addition, this central facility shall be capable of communicating with all of the nodes in order to assist in the detection of disturbances and to assist in the recovery of normal operations.

SECTION VI

CANDIDATE TEST BED ARCHITECTURES

6.1 INTRODUCTION

This section presents three different test bed architectures that will meet the requirements described in Section V. Each architecture fits into the basic configuration shown in Figure 6-1. A fullup test bed includes circuit and packet switching capabilities as well as the ability to communicate with a central network monitoring and control system. It is capable of expanding to a minimum of three sites. For each experiment, the configuration of the test bed would vary, but each variation is a subset of the basic overall configuration.

Common to all architectures is the central network monitoring and control center (NMC). This is identical for all test bed options. The NMC, shown in Figure 6-2, has as its major requirement the provision of network monitoring and performance assessment capabilities for the test bed. This requirement can be broken down into the following functions:

a. Statistical data gathering to monitor the performance of the test bed and present the results in terms of statistical summaries.

b. Observation of network status to assist in the detection of disturbances and recovery of normal operation.

The NMC comprises a PDP-11 type processor with the UNIX operating system.

The three candidate system architectures are shown in Figure 6-3 and are defined as follows:

Option I - An off-the-shelf commercially available circuit and packet switch as the basic components of the test bed. These switches require modification to satisfy the experimental requirements.

Option II - Discrete build-up of circuit and packet switch capabilities. Completely new hardware and software are used to satisfy test bed requirements.

Option III - Combination of Options 1 and 2 with Option 1 used for near term experiments and Option 2 for far term experiments.

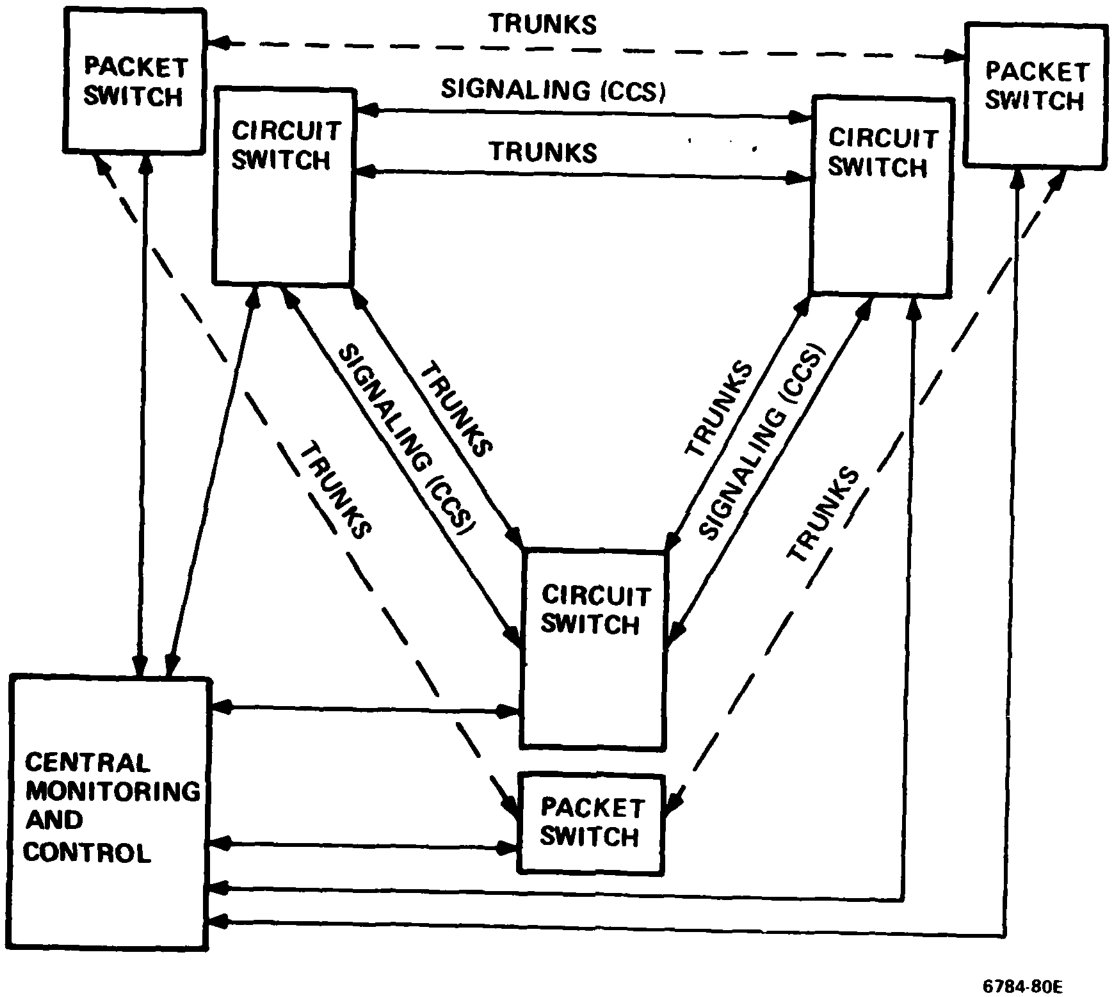


Figure 6-1. Overall Test Bed Network Architecture

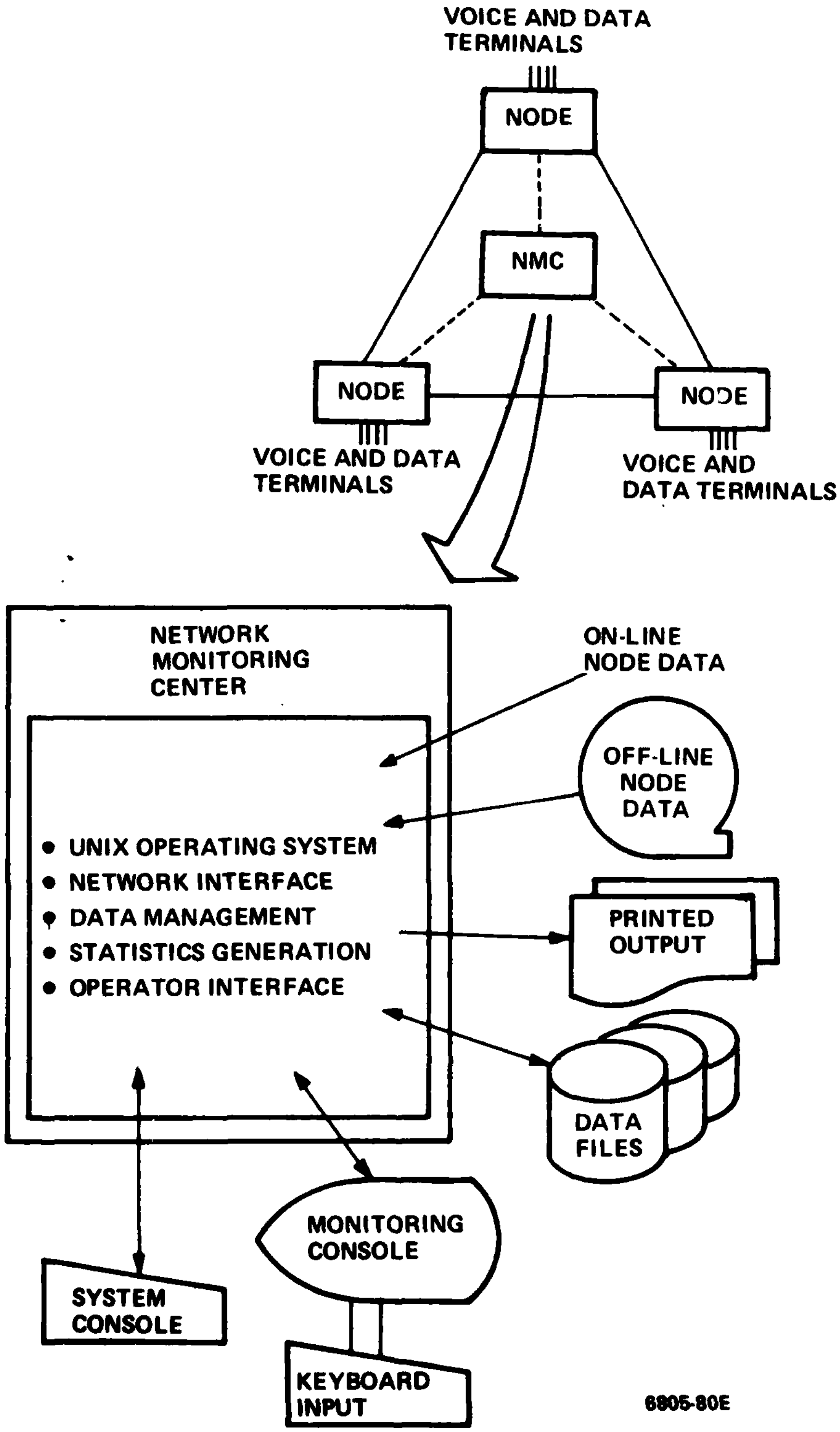


Figure 6-2. Network Monitoring and Control Center (NMC)

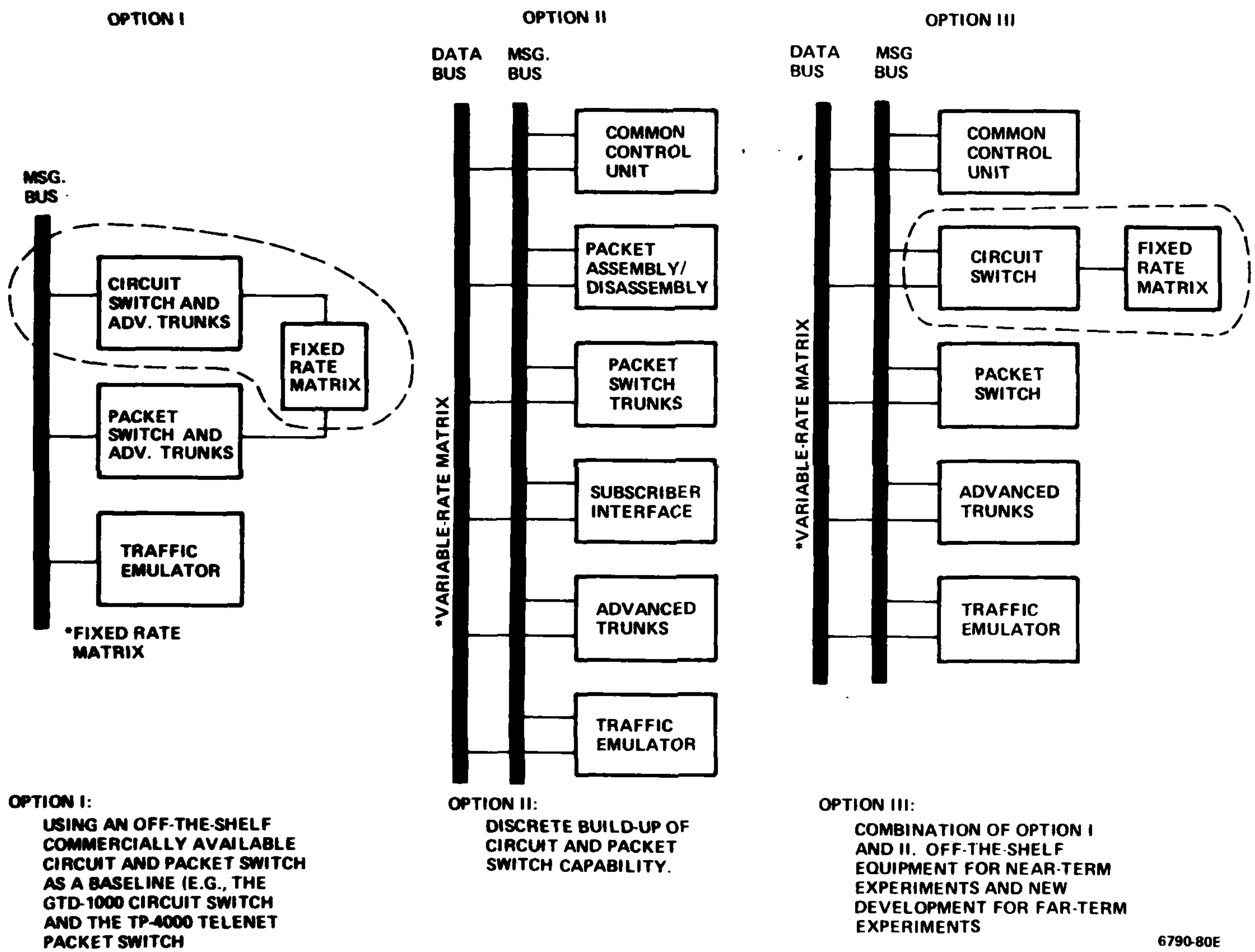


Figure 6-3. System Architecture

The advantages and disadvantages of these three approaches and the modifications required to perform the various experiments are discussed in the following sub sections. Figure 6-4 summarizes the advantages and disadvantages of each approach and identifies some differences in their capabilities.

6.2 MODIFICATIONS REQUIRED TO PERFORM EXPERIMENTS

The test bed requires several stages of development in order to perform the experiment groups designated in Figure 4-1. These stages are defined as:

Circuit Switching experiment groups (1,2,3,5,7,9,10,14)
Packet Switching experiment groups (4,8,15)
Hybrid Switching experiment groups (6,11,12,13,16,17)

In building the test bed to accomodate these experiments, we envision as a baseline a digital switch with commercial features and incremental software and hardware modifications as required. The modification necessary to perform the circuit, packet and hybrid experiments are discussed below.

Circuit Switching Experiments

These experiments emphasize interoperability and routing and require that features and interfaces be capable of being modified or added to the baseline switch for experimental purposes. The following changes are necessary to add military features to the baseline switch:

a. An addition of multi-level precedence and preemption (MLPP) including: 5-levels of precedence, Subscriber preemption, Trunk preemption, Preemption warning tone, and 4 by 4 keypads (DTMF).

b. AUTOVON, DMATS and FTS signaling additions. Minimal hardware changes are expected for this, since the signaling interfaces that exist for commercial equipment (e.g., E&M, CO, etc.) are principally controlled by software. Timing for winks, pulses, etc. are also software controlled.

c. Expansion of classmark capability.

d. Addition of tandem routing, alternate routing, least cost routing, and survivability routing.

e. Numbering plan modifications.

#	Description	Advantages	Disadvantages
Option I **Off-the-shelf Build-up**	Use of an off-the-shelf commercially available circuit switch and packet as the baseline. Added to this are military features and interfaces, combined voice and data capabilities, TASI, and advanced hybrid techniques.	• Reliable System • Low Cost • Immediately Available • Large Selection of Features • Selection of Interfaces • Diagnostics Available	• Cost and Risk of Learning Existing Software • Cost of Learning Hardware • No Military Features • Limited Architecture • Geared to 64 Kb/s Fixed Rate Matrix
Option II **Complete Discrete Buildup**	Discrete buildup of circuit, packet and hybrid test bed capabilities. This option requires all new hardware and software design.	• Custom-Designed Software and Hardware • Choice of Software Support Packages • Hand-picked Hardware • Virtual Matrix Structure • Systematic Growth and Evolution • Easily Modified Design • Variable Rate Matrix	• Architecture Not as Applicable to Near Term Experiments • High Risk/Cost for New Software/Hardware new Development • Feature Ability May Have To Be Reduced Because of Cost • Start-Up Costs for Hardware/Software Design • Hardware Designed Simultaneously with Software
Option III **Combination Test Bed**	Combination of Option I and Option II. An off-the-shelf circuit and packet switch is used for near-term experiments. Advanced concepts require a new matrix and new hardware and software. The near Term test bed evolves to become the access area for the full-up test bed.	• Parallel Development • Less Risk and Cost for Near Term Than Option II • Hardware For Far Term Can Be Developed Early • Variable Rate Matrix	• Merging of Near-Term Equipment Into Far-Term Is Expensive • High Risk/Cost to Modify Existing Software • Cost of Learning Software/Hardware • Possibility of Multiple Software Languages

Figure 6-4. Candidate Node Architectures

Experiment Group 5 requires the addition of voice and data on a D2/D3 T1 channel while Experiment Group 7 requires the use of TASI on a T1 channel. This necessitates the design of Silence detectors, TASI map handlers, and Multi-port multiplexers. Circuit experiments involving Common Channel Signaling (Experiment Groups 9 and 10) require a dedicated serial line interface and software to implement a modified CCITT#7 (i.e., CCITT#7 with the addition of military features).

Packet Switching Experiments

Packet switching experiments assume a standard packet switch configuration as the baseline with the following modifications:

a. Addition of multi-level precedence and preemption.
b. Addition of special features such as multi-addressing.
c. Traffic load capability.
d. Statistics generation.
e. Special routing capability, e.g., PAR, PVC, mixed routing.
f. Interface to the circuit switch.

Experiments involving packetized voice (Experiment Group #8) require special hardware/firmware to perform voice packetizing and depacketizing. Advanced data experiments involving various protocol and interface designs (experiment group #15) also require hardware and software additions.

Hybrid Switching Experiments

The baseline for hybrid switching is a combination circuit and packet switch. The following additions to this baseline are required in order to perform hybrid experiments:

a. Capability for a flexible bandwidth trunk.
b. Automatic test equipment.
c. A network monitoring and control facility which would communicate over a CCS channel.
d. Recovery capability.
e. Hybrid software enhancements to the call set-up, routing, and network control programs.

Combined voice and data network control requires the capability to send CCS information via the packet switch as well as packet network information and status by CCS.

The ability to have fixed bandwidth and variable boundary capability requires hardware. A T1 trunk will have to be developed that can run at various framing rates and provide variable bandwidth slot assignments. Software in the circuit and packet switches must be enhanced to manage the dynamic contention for trunk bandwidth.

6.3 OPTION I: USE OF OFF-THE-SHELF, COMMERCIALLY AVAILABLE EQUIPMENT

The approach for the first candidate architecture is to use an off-the-shelf, commercially available circuit switch as the baseline. Software would be modified, rewritten or replaced, and new hardware interfaces would be built as necessary to allow the circuit switch to perform the functions of a test bed. Later, an off-the-shelf, commercially available packet switch can be added to the test bed. The packet switch also requires modifications to perform the various experiments.

The baseline circuit switch is a program-controlled digital PABX, with time division switching utilizing standard commercial Pulse Code Modulation (PCM) techniques. This digital switch (e.g., a GTD-1000) contains a full set of commercial features that provide a good foundation for the features required in the test bed.

The packet switch is a commercial packet switch data communications system that will enable communication with a wide variety of computers and data terminals. A typical representative is the Telenet TP-4000. This system has the ability to interface with X.25, HDLC, Bi-synch, ADCCP and SDLC systems.

Advantages of using off-the-shelf commercially available equipment include the following:

a. The use of existing developed hardware is inexpensive compared to developing new hardware for the test bed.

b. The hardware is reliable, readily available, and provides a definite low risk factor.

c. Commercial DDD interfaces are available including D2/D3 T1 interfaces.

d. The digital matrix is representative of digital switches using standard PCM.

e. Diagnostics are available.

In general, Option I offers a known, immediately usable package from which to initiate experiments.

Disadvantages of Option I become apparent as soon as changes have to be made to the system. The actual changes required for experiments are discussed in detail in Section VII. However, a summary of the disadvantages of using off-the-shelf switches for the test bed is provided here:

a. Detailed understanding of the software is required before a single instruction can be modified. In general, the existing software for a circuit switch or packet switch is developed for a product line and is not designed to be modified by the user. The detailed software knowledge required to modify existing software is thus a major start-up cost in Option I. An alternative is to rewrite all the software using existing hardware. However, this apprroach is also costly. A third alternative is to let the original developers of the software package make the changes. This, generally, is not acceptable to the switch vendor due to their own manpower limitations, and the proprietary nature of vendor software.

b. Hardware must be learned sufficiently well before modifications are made for the test bed.

c. In order to modify any software, facilities for assembling and linking the programs together are required. This includes the following support software and development tools: Assembler, Compiler, Linker, Simulator, and Absolute listing program.

d. Some off-the-shelf switches require programs for generating the database to be used by the switch. Such programs must be learned and modified to accomodate test bed requirements.

e. Off-the-shelf circuit switches and packet switches have fixed architectures that may not be flexible enough for the advanced concepts.

f. Variable bandwidth, variable frame, and moveable boundary hybrid trunk experiments also require extensive hardware changes. Trunk circuits will have the task of synchronizing variable bandwidth assignments. The fixed bandwidth nature of the circuit switch presents problems in that extensive multiplex timing and buffering has to be done at both ends of a switch matrix connection. Delays from input to output may be distorted due to the additional buffering.

Architecture for circuit, packet and hybrid switching experiments under Option I are shown in Figures 6-5, 6-6, and 6-7, respectively. Figure 6-8 identifies some of the software functions associated with the approach. The architecture for the circuit switching experiments,

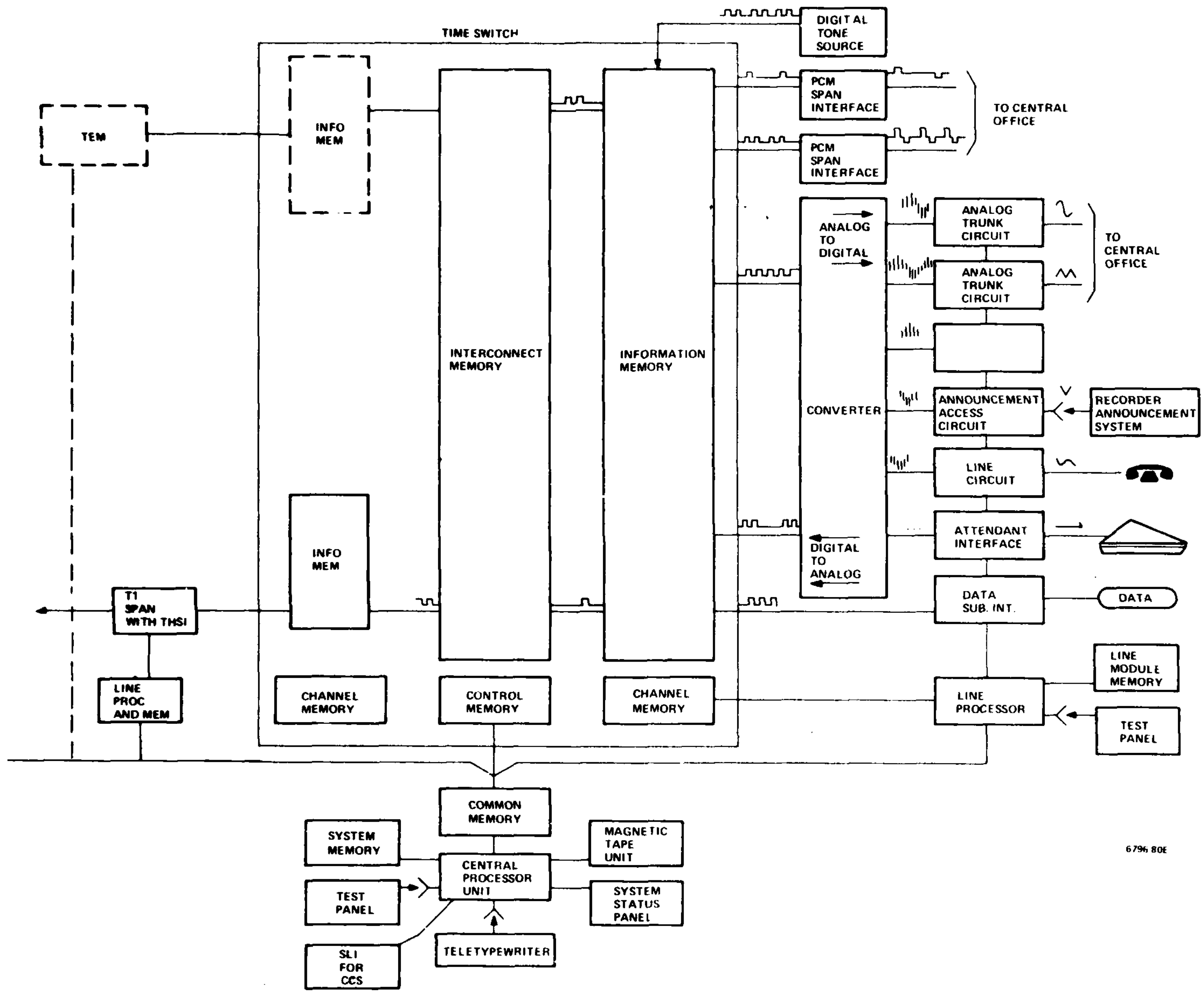


Figure 6-5. Option I Architecture for Circuit Switching Experiments

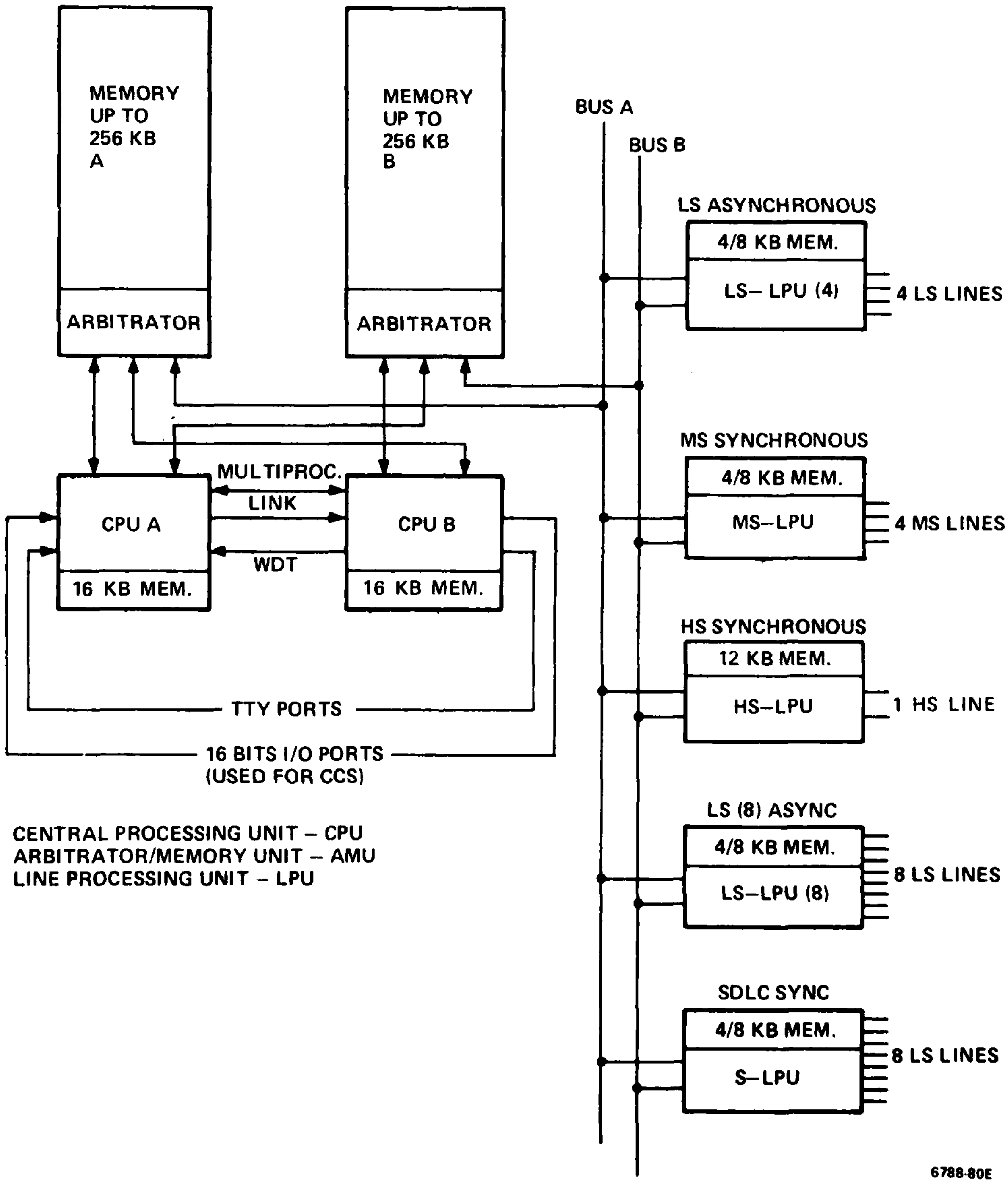


Figure 6-6. Option I Architecture for Packet Switch Experiments

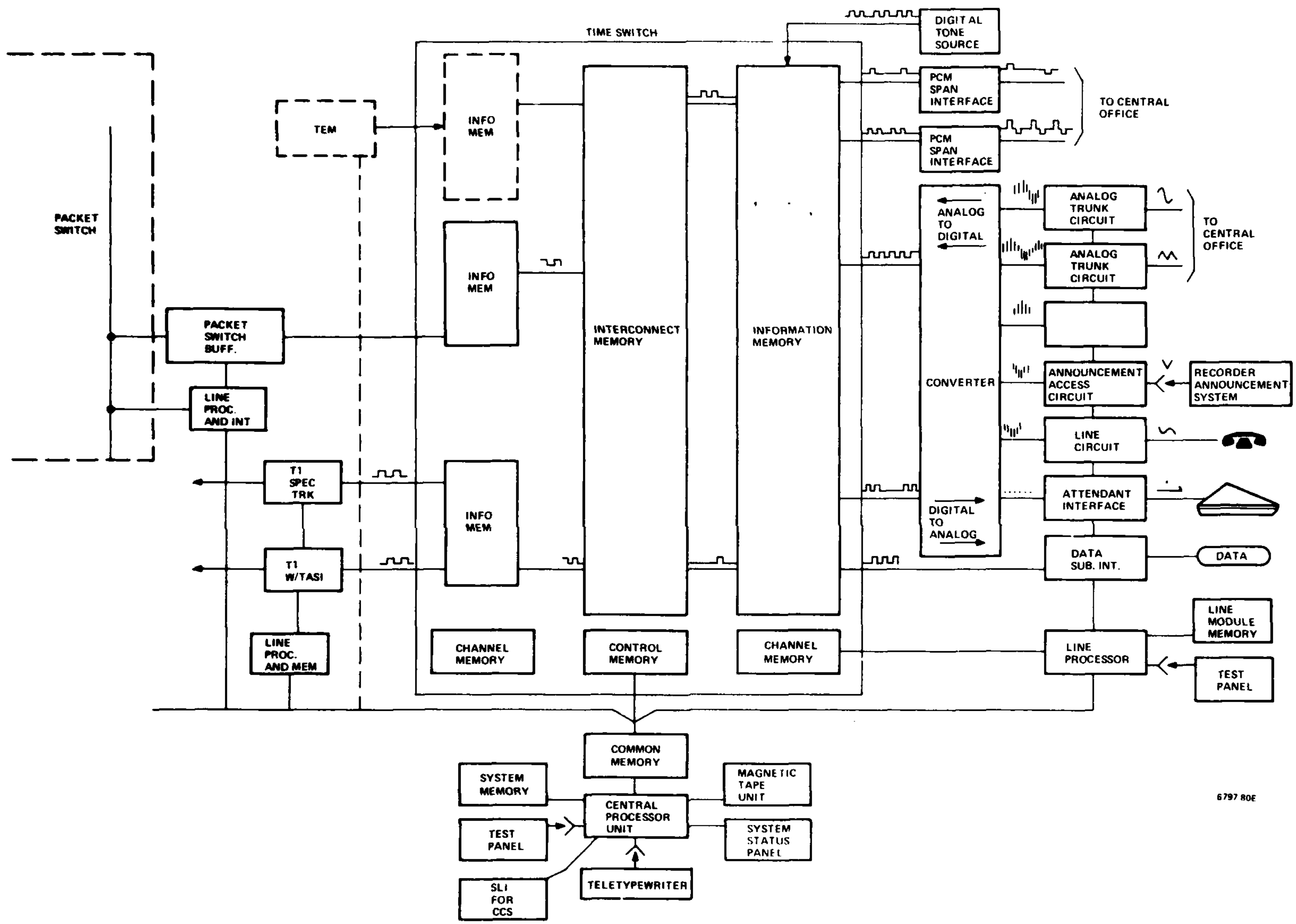


Figure 6-7. Option I Architecture for Hybrid Switching Experiments

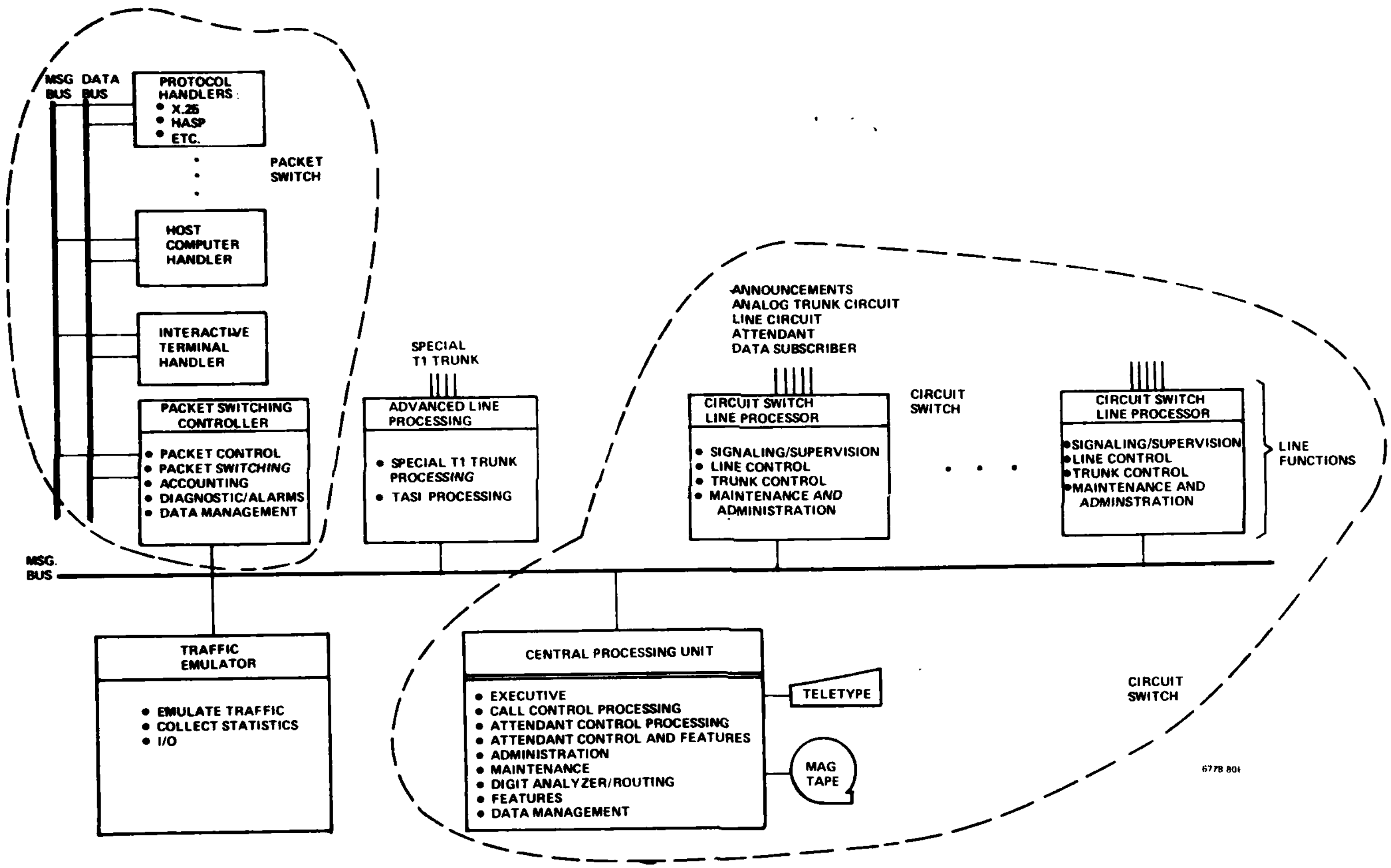


Figure 6-8. Software Functions for Option I Architecture

shown in Figure 6-5, is the typical structure of a modern software controlled digital switch. This architecture has a time division matrix, geared to multiples of 64 Kb/s channels (each channel consisting of 8 bits at an 8 KHz sample rate). Data is transferred between A/D converters associated with each analog terminal and trunk. Line circuits are controlled and scanned by the line processor. Each line processor handles several line/trunk circuits for such functions as dial-pulse collection, seizing, hook flash, ringing and general interface sequencing.

Digital subscriber and trunk interfaces bypass the A/D converters and tie directly to the digital matrix. Their supervisory functions are controlled in much the same manner as the analog lines and trunks.

The central processor interfaces and controls several line processors and the matrix. It has common function responsibilities such as routing, matrix control, data base management, I/O, CCS (Common Channel Signaling), Sender/Receiver assignment and other general purpose functions.

Figure 6-5 also shows a TASI T1 trunk interface. This would be either a new or a modified design of a standard T1 interface, with the principal differences being the number of input time slots that can be assigned and the TASI map control. This type of T1 would therefore have to replace at lease two standard T1 functions in an existing system.

Traffic emulation is shown in dotted lines since it need not be a separate function if internal software is used to generate traffic. If external traffic generation is required, the emulator must be added in the form of an additional line processor with special call emulation capability.

The architecture for Packet Switching experiments under Option 1 is shown in Figure 6-6. It is a distributed processing system architecture, in which various line and trunk interfaces have their own special processors, buffers and bus interfaces. All level I and level II protocol is handled by these line processors. Another processor,

the CPU, controls routing and level III protocol, and is also responsible for inter LPU communication, i.e., setting up the transfer path of data from input to output. The main memory associated with the CPU contains both data base and temporary buffering. This serves as a common buffer area for overlow data that cannot be stored in the short temporary buffers of the line processing units.

Hybrid switching experiments for Option 1 assume the configuration shown in Figure 6-7. This configuration uses the packet and circuit switches described previously but requires development of new interfaces for the new trunk types and development of a packet to circuit switch data base interface. The two new T1 trunk interfaces appear as additional line processing units with direct ties to the matrix. Much of the special processing for trunk control and mapping is handled by the special line processor that "translates" trunk protocol to something the central processor can understand. Difficulties in processing trunk data include the variable bandwidth characteristics of individual slot assignments and the moveable boundary between switching regions. Call setup/take-down times and synchronization with the matrix are critical to avoid buffering at the trunk interface. In the more complex variable bandwidth, variable boundary trunk case, the matrix interface becomes even more critical. The fixed bandwidth nature of the matrix (64 Kb/s increments) makes it very inefficient for lower rate data transfer. Bandwidths that are not exact submultiples of the 64 Kb/s are extremely difficult to handle. Even for exact multiples of 64 Kb/s there are still difficulties in maintaining internal synchronization for the correct number of samples.

Connection to the packet switch is also through a special T1 interface to the circuit switch. The line processor in this case serves to interface the central processor bus of the circuit switch to the CPU bus of the packet switch. Data is stored in a special buffer that is synchronous to the circuit switch matrix timing. The disadvantage with this approach is the vairable bandwidth nature of the advanced integrated trunks. Buffering must be extended to trunk interfaces to accomodate the incremented bandwidth nature of the matrix.

6.4 OPTION II: COMPLETE DISCRETE BUILD UP

The second candidate test bed architecture is a completely new system build up. This architecture is distributed and designed specifically for this application. Knowledge gained from other programs [1, 2, 3] has been applied where applicable, however, the architecture has been driven mainly by the requirements of Section V. Main features of this system architecture include: a) structured higher level programming; b) hand-picked processors and hardware; 3) custom designed software; d) selected software support packages.

Structured and modular programming is of particular importance to a test bed because of its generic requirements. Programs are likely to change often as new developments and ideas arise. In most systems, once a program is working, there is little incentive or need to change the program. In designing a production system, the interactions and limitations are well defined. This is not the case for a test bed. One can expect redefinition of entire program areas such as routing depending on the experimental requirements. It is, therefore, extremely useful for different functional areas of programs to be modular.

Custom built hardware for a totally new test bed architecture also has distinct advantages. The matrix structure more closely resembles that of a virtual matrix [2]. This would match closely the variable bandwidth characteristics of the packet switch and the slot assignments on the advanced T1 trunks. Special consideration can be given to handling the inter-processor communication needed for the packet and circuit switch functional interfaces of the integrated switch. Systematic growth and evolution is more easily accomplished. A further advantage of designing from "ground up" is that portions of the system subject to variation such as interfaces can be specifically designed for modification. This would lower the costs of experimentation in later experiments.

Disadvantages of this approach concern mainly the near term experiments which involve investigation of deficiencies or limitations in existing equipment. Areas of experimental study will not generally be concerned with pushing the state-of-the-art so much as with system

protocol and procedure problems. As such, a high technology solution and a bottoms-up approach is more costly and presents a higher risk than would seem necessary for these experiments. In addition, the cost and development schedule to establish full "POTS" operation with all the assumed features would be high. Because of this high cost, features and operation for this approach should be reduced to the bare minimum necessary to run the experiments.

Start-up tasks under this option include the following developments:

1) Basic switch architecture
   a. matrix and control
   b. internal data and control mechanism
   c. program loading and I/O capability
   d. processor elements
   e. subset interfaces (DTMF, Dial Pulse)
   f. trunk interfaces (POTS)
2) Software System
   a. high order language
   b. support software
   c. development system

The hardware architecture proposed for the completely new build up test bed is based on a distributed system architecture (see Figures 6-9, 6-10, and 6-11). Software functions are shown in Figure 6-12. Principal characteristics are the separate message and data bus systems. The data bus is specifically designed to handle variable bandwidth slots. It operates under its own synchronous control. Each interfacing module can have its own bandwidth based on its requirements. The data bus adapts the data transfer rate for that module to be compatible to its requirements. Overall bandwidth of the data bus can be designed to a range of several megabytes per second with the bandwidth per call adjusted to meet interface requirements. Typically, the bus handles ten 64-Kb/s calls with the same bandwidth as five 32-Kb/s calls. This differs from a standard circuit switch in that bandwidth is based on the number of calls and not the speed requirements. The data bus is roughly equivalent to an adaptive multiplexer.

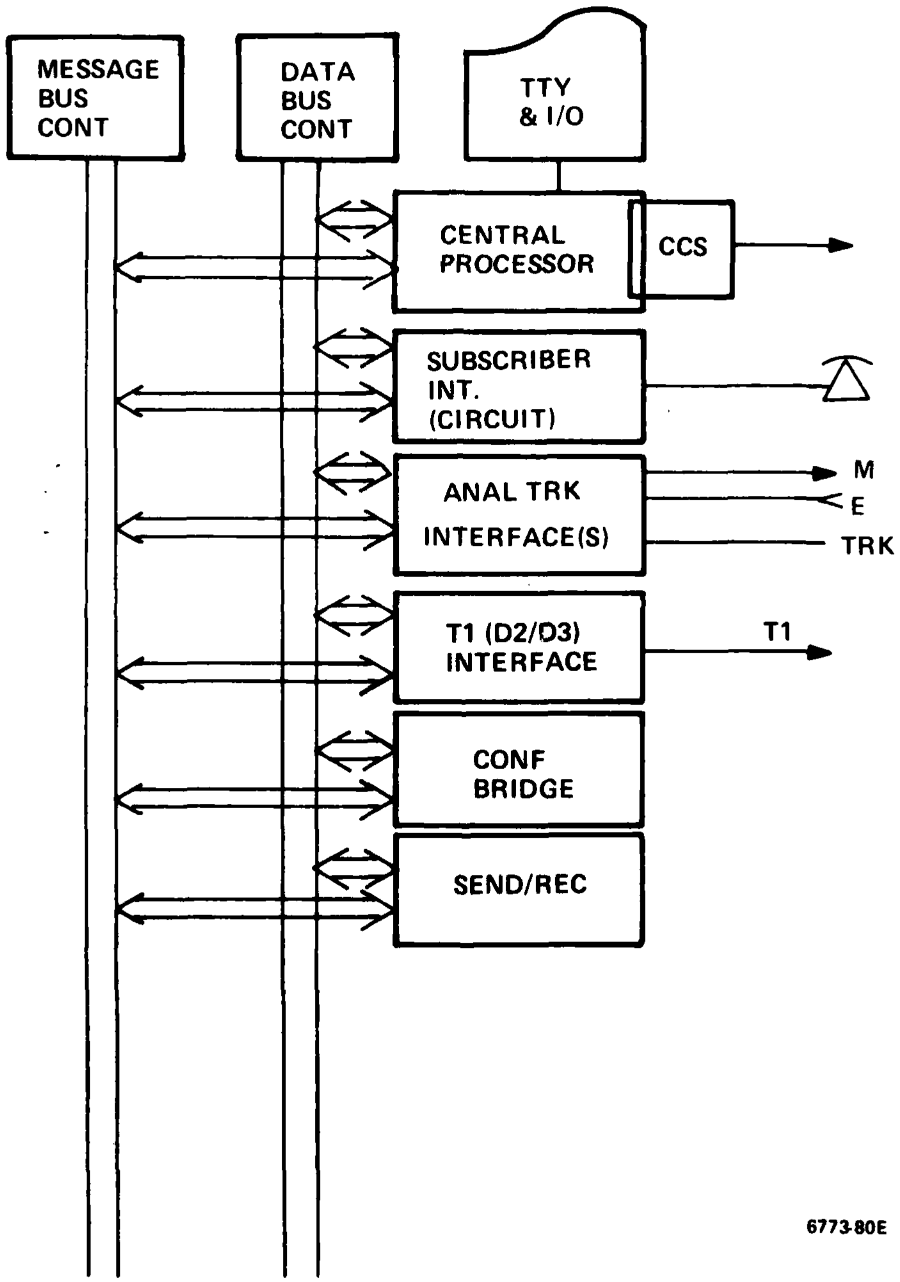


Figure 6-9. Option II Architecture for Circuit Switching Experiments

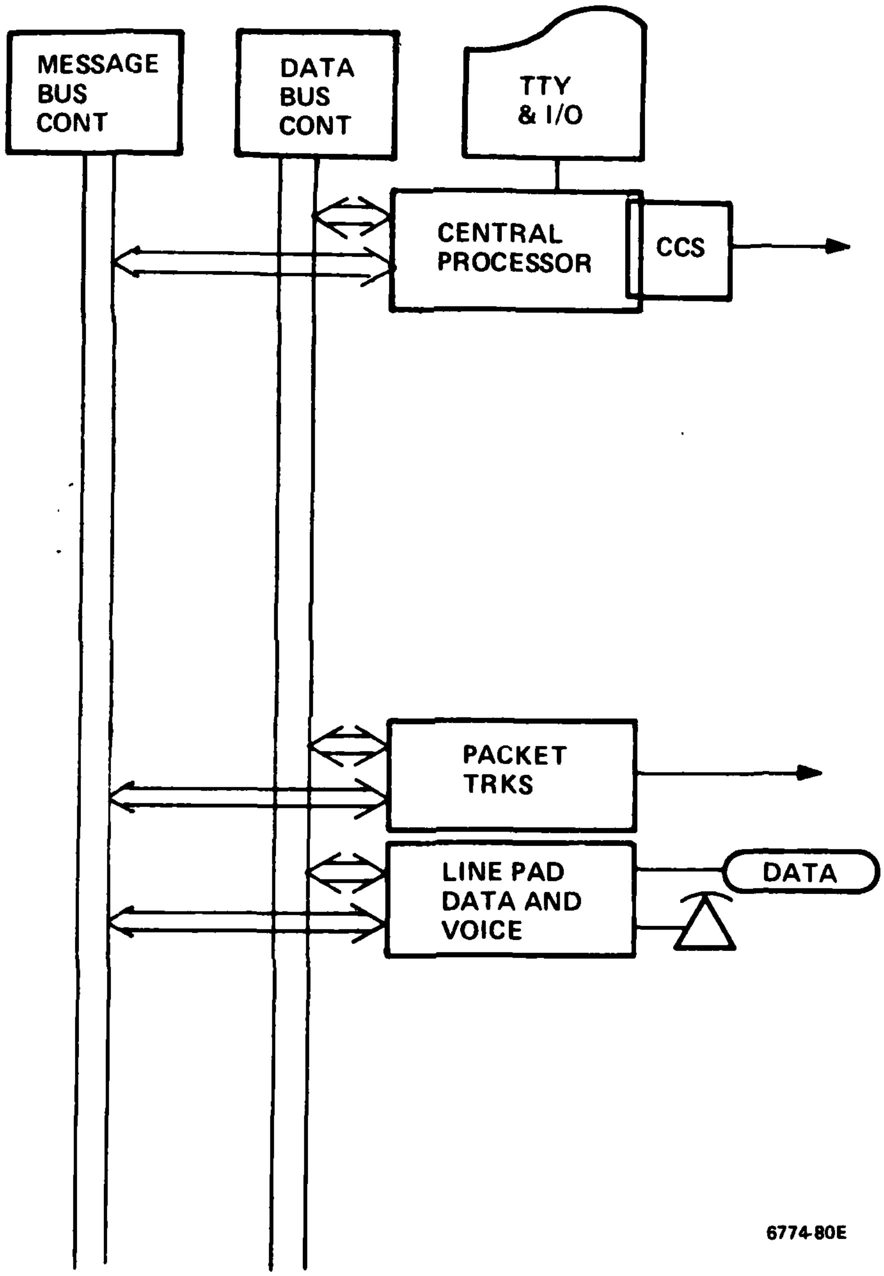


Figure 6-10. Option II Architecture for Packet Switching Experiments

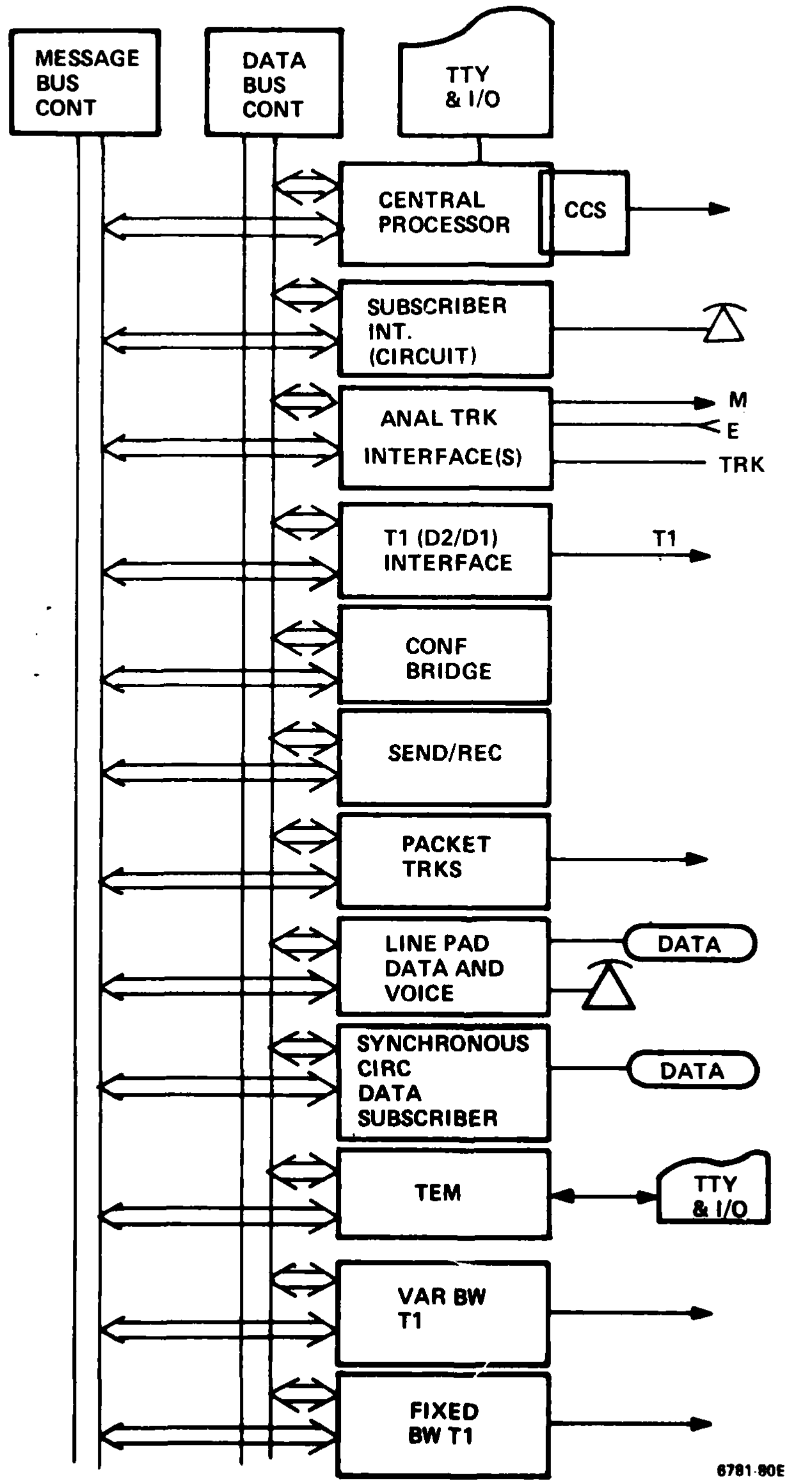


Figure 6-11. Option II Architecture for Hybrid Switching Experiments

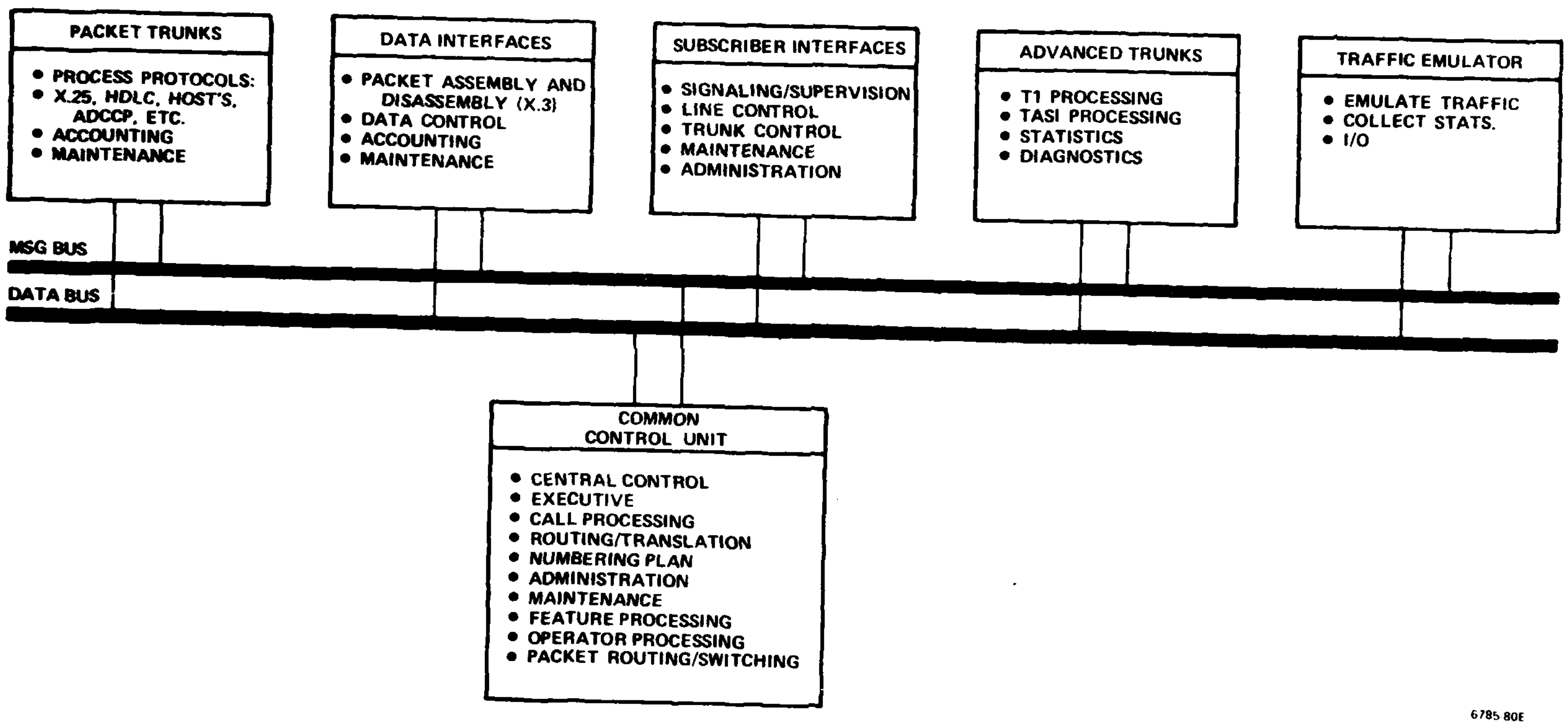


Figure 6-12. Software Functions for Option II Architecture

The message bus and communication path between the different modules is similar to that of a standard distributed processor system. Messages are passed on a queued or time available basis.

Functional blocks for each of the interfaces such as lines, trunks, sender/receivers, etc. have their own processing control which is responsible for the supervision and control of the individual interfaces. The routing, data base and general system contol functions are handled by the Control Processor.

Packet Switch functions (shown in Figure 6-10) are an extension of the distributed system architecture. The Line PAD functions for voice and data and the packet trunks handle the Level 1 and Level 2 protocols. Data and message busses operate identically to those of the circuit switch. The Control Processor has similar functions to the CPU of Option 1 and to the circuit switch processor of Figure 6-9. In fact, the same processor can be used with a new program.

The hybrid configuration of Option II (Figure 6-11) is formed by adding modules to the basic bus structure. Advanced trunks are treated as new interfaces. They contain their own specific processing to handle Level 1 and 2 protocols. This includes map control and boundary movement. Control programs that combine both the circuit and packet CPU functions of the hybrid, can either be located in a single central processor as shown or can be designed as multiple CPU functions attached to the bus. The TEM would be treated as a separate module(s) and added as required.

6.5 OPTION III: COMBINATION TEST BED

A third candidate test bed architecture is a combination of the first two options. In this alternative the near term experiments are run with an off-the-shelf digital switch as the baseline. This makes maximum use of existing technology and current design in order to run experiments on present-day system problems. The circuit switch used for these experiments then becomes the access area in an expanded test bed. Advanced concepts are evaluated in this expanded test bed using the new matrix and hardware. The packet switch is a compromise between existing hardware/software/firmware and new development.

Software development for the existing equipment would track the approach of Option I, i.e., either a modification of existing software or a major re-write. The direction depends on the particular switch chosen as the baseline.

Advantages of an approach which combines the first two options are:

a. The potential for parallel development of different portions of the test bed.

b. Schedule and risk for the near term experiments are improved over that of the distributed approach.

c. Hardware development for the advanced condepts can be started considerably before the software is required.

The disadvantages of this approach are:

a. Modification of existing software for near term experiments presents some risk.

b. The possibility of multiple software languages for the two approaches.

c. The merging of existing equipment into an advanced system concept could present some problems.

The principal attraction of this approach is the decreased schedule and risk for performing an entire set of experiments covering both the near and far term. Advanced concepts require several key areas of development. The new trunks require extensive development to establish the feasibility of different formatting schemes. Overall network concepts and derived experiments assume that these trunk formats are possible. In a combined test bed, development of these trunks can begin early and in parallel with the earlier stages of experimentation. In fact, much of the advanced concepts experiments can be performed concurrently with the set of near term experiments dealing with the present equipment and network.

The hardware architecture is a combination of the first two options. Architectures for the circuit and packet switch experiments (see Figures 6-5 and 6-6) are identical to those presented for Option I. These are based on off-the-shelf digital switches modified as appropriate.

The major difference is in the architecture to perform the hybrid experiments (Figure 6-13 and 6-14). Here the approach is to combine the variable bandwidth distributed architecture presented for Option II with the circuit and packet switches of Option I, the latter acting as access functions to the former. Advanced concept trunks with variable boundaries, new central processors, and multi-rate subscribers are attached to the hybrid bus. Packet subscribers and 64 Kb/s PCM subscribers obtain access through their respective switches. The functions of the central processor are to control routine and provide other common controls for the node. Routing is performed for both the network and inner-access area levels. The node thus becomes a hierarchy of switches with packet and circuit switches acting as access to the higher order distributed architecture which provides the tandem function.

In order to tie an existing off-the-shelf digital switch to the variable rate bus, some modifications are necessary. The circuit switch requires a timing buffer for short temporary storage of the data so that the data and message buses can run independently. Interactive processing with the message bus is performed through a "protocol" conversation processor that allows a standard circuit switch to interface as a modified trunk and at the same time allow the hybrid message bus to use its standard protocol. A similar capability is also required for the packet switch. Here, the single computer data/message bus must be separated to interface with the two buses for the distributed system. Local buffering is required to match the batch data transfer of the packet switch to the fixed variable rate synchronous nature of the distributed system bus.

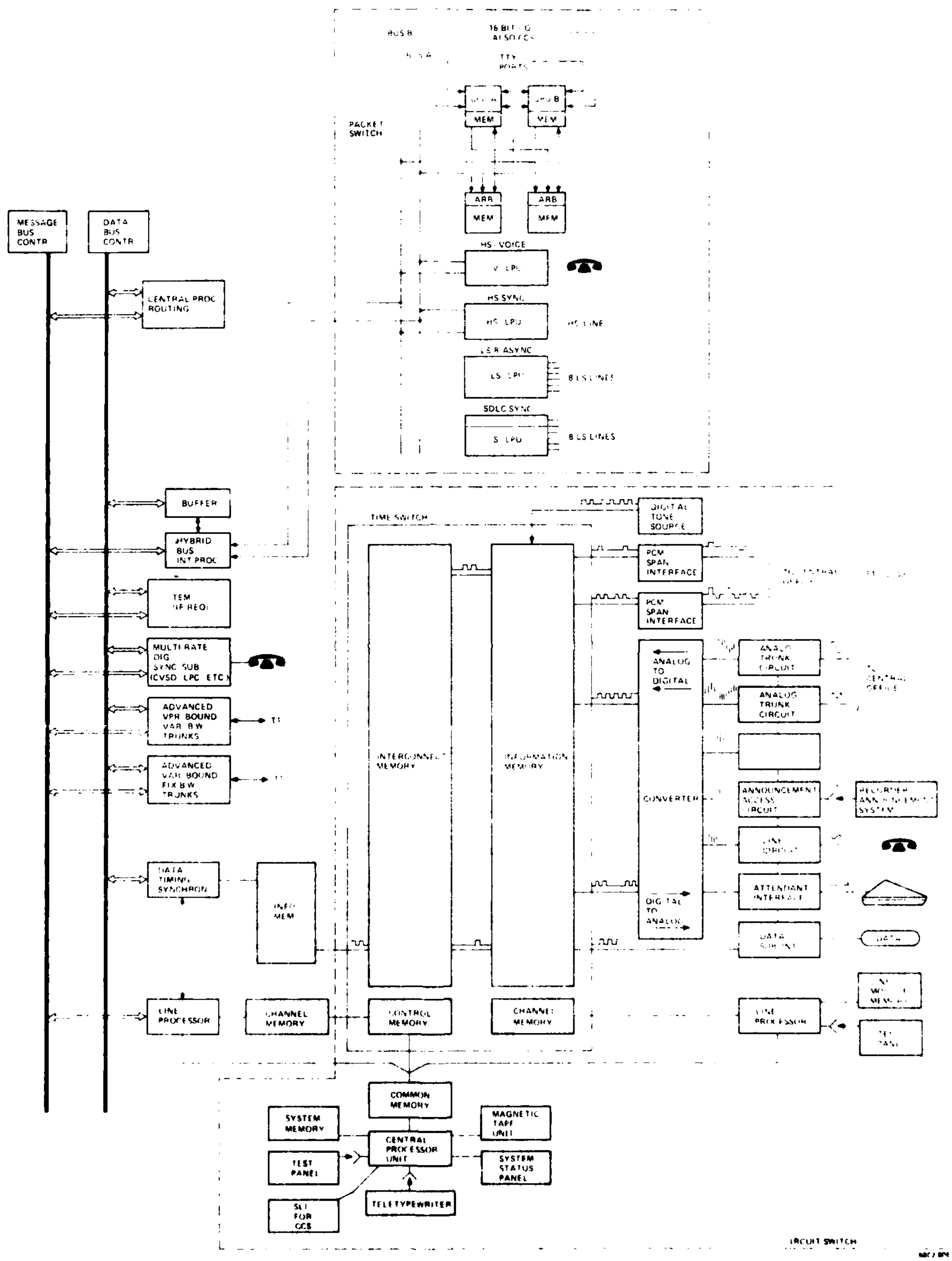


Figure 6-13. Option III Architecture for Hybrid Switching Experiments

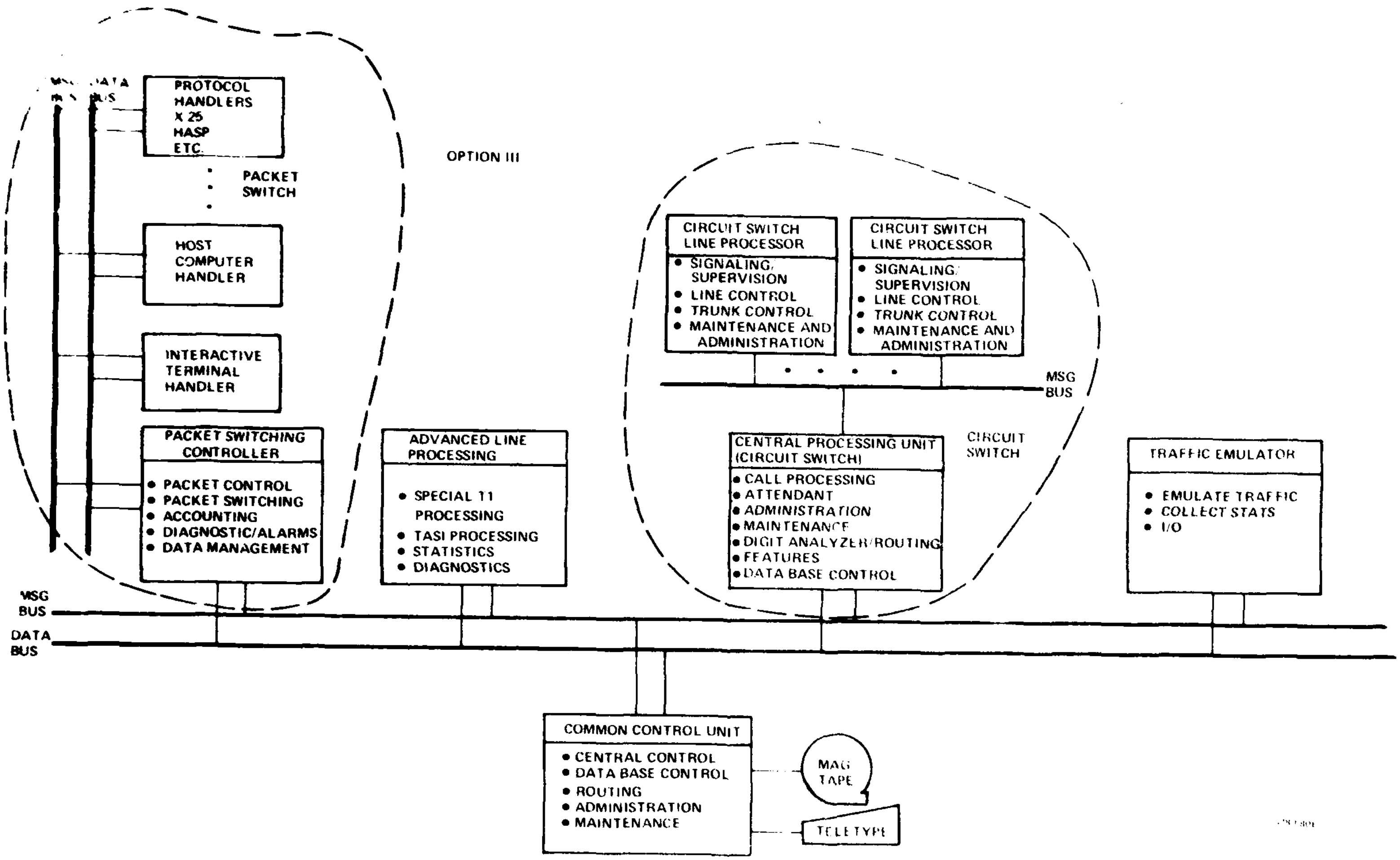


Figure 6-14. Software Functions for Option III Architecture

Other interfaces tied directly to the hybrid buses include digital synchronous voice and data subscribers not requiring a 64 Kb/s PCM rate and the TEM interface. The principal advantage of this type of architecture is a "best fit" approach to the required tasks to be performed. The circuit and packet switching experiments employ today's technology and equipment and reflect modern fielded systems. The Hybrid switching experiments allow future concepts to be tested with test bed equipments designed specifically for that purpose. This architecture also allows the three areas to develop and proceed independently for a considerable period of time before they are required to be integrated.

REFERENCES FOR SECTION 6

1. Electronic Telecommunications Switching System (ETSS), A family of custom-built military and commercial analog switches presently in operation in various parts of the world.

2. Digital Switch Virtual Matrix, IR&D program 993-1905, 1977-1979, Application of digital matrix techniques to voice and data switching.

3. SENET-DAX Study, Final Report (Volumes I & II) and Supplemental Study Report, DCA Contract 100-75-C-0071, 1975-1976, Defense Communications Agency study on the integrated switching and multiplexing of voice and data in a hybrid system.

SECTION VII

TEST BED CAPABILITY/COST/SCHEDULE PROFILES

7.1 INTRODUCTION

This section will present the cost and schedule trade-offs for specific implementations of the generic architectures presented in the previous section. These implementation approaches, although specific, are felt to be representative of the general classes of equipment architectures in the field today. Option I (Off-the-Shelf) is a combination of several GTE equipments. The GTD-1000 is used for the circuit switch and a Telenet TP-4000 system is used as the packet switch. Option II is a discrete buildup in hardware and software of a new distributed architecture. It uses distributed micro-processors and makes use of higher order languages in its development. The third option is a combination of the first two approaches. It makes use of off-the-shelf equipment for the early experiments and then utilizes this equipment as the access functions of a distributed hybrid switch architecture. A more complete description of the three switch equipments, i.e., the GTD-1000, TP-4000, and discrete buildup, is included in Appendices A, B, and C, respectively.

In performing the tradeoffs, there were several assumptions made that affect the costing. These are:

a. Costing does not include studies or time to perform the tests. The data reflects the cost of implementing the test bed once the problem has been defined

b. Steps are taken sequentially so that things specified in an earlier step are assumed to be already implemented at a later step. This precludes a massive reordering of experiments while maintaining the same cost data

c. Documentation and acceptance testing are in accordance with commercial grade practices. Commercial grade documentation and brass board construction are felt to be sufficient for a test bed

d. Equipment is available for development. This assumption requires that the test bed be available to the developers during the build phase to test new changes and integrate new programs

e. Vendors are cooperative. The provision of vendor documentation and a vendor training course for both software and hardware is essential for the off-the-shelf equipment bids. A vendor staffed test bed team is not considered nor is cost of courses from the vendor.

In addition to the 17 defined experiments, there are three additional costing steps. These represent the baseline costs associated with establishing the basic capabilities for circuit, packet and hybrid switching. They are costed out separately so that any particular experiment does not unduly bear the major cost for their development.

Cost results are presented in several ways. The first is for a step-by-step sequential buildup. This results in a lengthly schedule for all three options. We next looked for methods to compress the schedule; however, the ground rules were still the same in that all experiments were required to be performed and they were to be done in order. The first method employed maximum concurrency. This allowed experiments to be done in parallel, assuming that the necessary equipment could be made available. In general, Options I and III allowed circuit and packet experiments to run concurrently prior to performing the hybrid experiments. Option II was not very suitable for concurrent implementation.

Another method aimed at reducing the schedule by allowing not only concurrency but also grouping of experiments. For example, experiments 1, 2, 3 and the circuit switch baseline were lumped into a single step. The result of this procedure was fewer steps and hence a shorter schedule.

The net result of compressing the schedule was still not satisfactory; therefore, further optimization based on changes to the literal interpretation of the test bed phased buildup was performed. The resultant approach is a dual test bed variation of Option III and is presented as the recommended approach in Section VIII. Costing for the recommended approach uses all of the same assumptions made in this section with some minor exceptions concerning experimental sequence. Prime cost data for the recommended model is extracted from the three options presented in this section.

7.2 COST, CAPABILITY, SCHEDULE PROFILE

This section outlines on an experiment by experiment basis, the costs, schedule and capability of each step. The hardware and software modules that have to be changed are listed. Each step includes a description of the general approach and task of that experiment and a diagram of the test bed configuration at that step. The hardware tasks include such things as new card designs, system assembly, drawing packages, and new equipment buys. The software task list includes new and modified software functions. A total of the estimated instruction count is included. As a final summary, the total cost and schedule of that step will be given for each option. The assumption for all steps is that the step before had been completed and all steps are performed.

7.2.1 Circuit Baseline

System Description

The circuit baseline step is included to establish the "get-started" costs of the three test bed architectures. The baseline is considered to include training courses, software development systems and a single-node, digital, circuit switch capable of the following:

- Standard DTMF
- Commercial T1 Trunk
- Call Forwarding
- Conferencing
- E&M Trunk
- Basic Routing
- 5-Digit Numbering Plan

Any features above this capability are considered extraneous and are only included if they come with no cost to the test bed.

Option I	Option II	Option III
Hardware Tasks:		
GTD-1000 (1PEC, 1CEC)	Matrix Bus	(Same as I)
Utility System (debug)	Message Bus	
Training Course	Main Processor	
Software Development System	DTMF SR	
Data Base Generator (CPG)	5 Party Conference	
	Subset Interface	
	T1 Trunk D2/D3	
	Operator	
	E&M Trunk	
	Software Development System	
	Circuit Switch Access Processor	
	I/O	
Software Tasks:		
Software Training Course:	Nodal Processor S/W	(Same as I)
System Level Course	Circuit Access S/W	
CEC Course	Other Software	
PEC Course	S/W Task Management	
Utility Course		
Install Support Software		
Data Base Generation (CPG)		
Configuration Control		
In-House Training		

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	6	145	86	- -	
II	16	154	310	16.2	
III	6	145	86	- -	

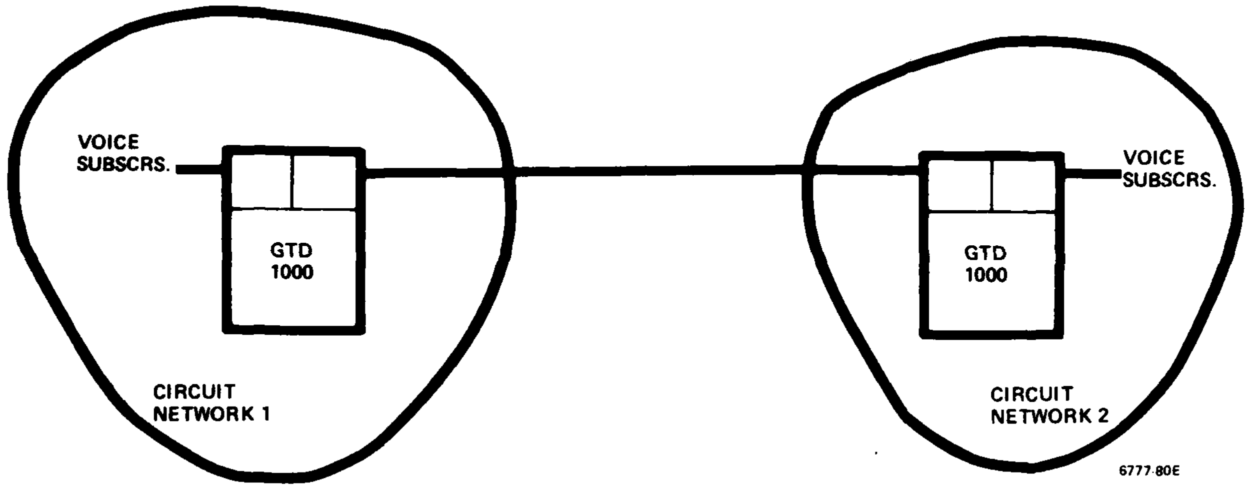
7.2.2 Experiment #1 - Voice Net k Interfaces

System Description

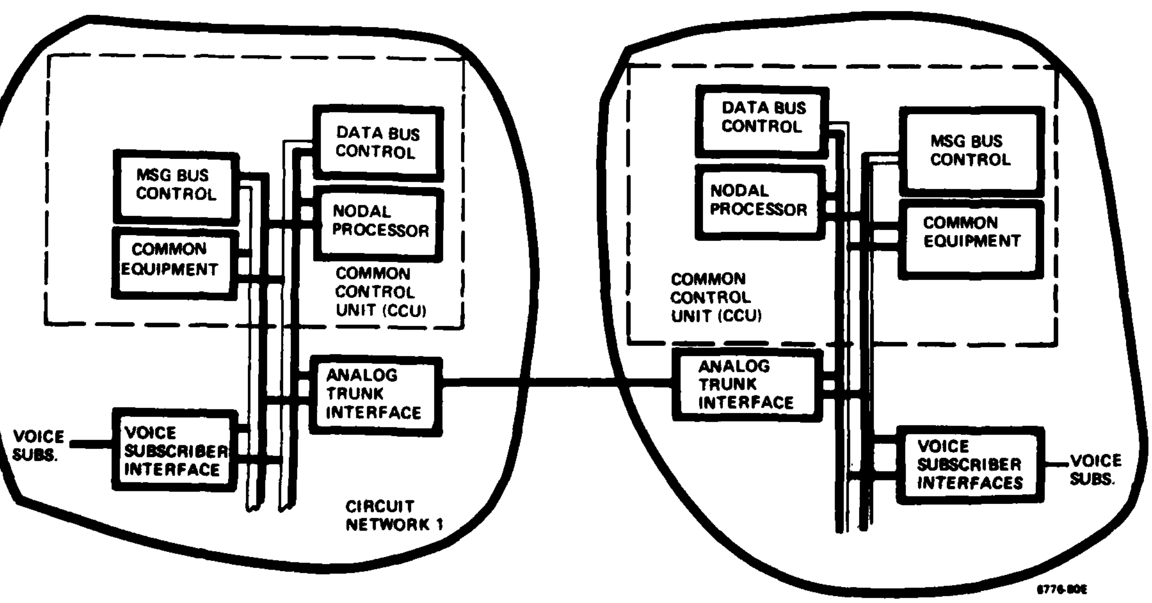
The test bed will require the addition of a second node to perform this experiment. It will also require the addition of military features such as MLPP. Three new interfaces are deemed necessary for the experiment: Autovon, DMATS, FTS. The numbering, routing and signaling plans must be modified to allow for class mark and feature restriction by using techniques such as digit insertion and number substitution. The signaling sequences for the new interfaces are assumed to be standard commercial hardware with software timing and sequence variations. This allows external commercial hardware adapters to convert E&M signaling in Option II. Option I & III have several commercial interfaces available as standard equipment.

Option I	Option II	Option III
Hardware Tasks:		
Second GTD (1PEC, 1CEC)) 4x4 Keypad DTMF New MIL Service Tones New Interface Changes MF REC	External Adapter MF S/R 1 More Node Added Service Tones	(Same as I)
Software Tasks:		
Interface Additions: AUTOVON, DMATS, FTS Routing Additions MLPP Modifications CPG Modifications	MLPP Numbering Plan Interface Additions: AUTOVON DMATS FTS	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	10	189	145	8.1	7-1a
II	7	91	100	5.5	7-1b
III	10	189	145	8.1	7-1a



(a) Option 1 & 3



(b) Option 2

Figure 7-1. Experiment 1 - Voice Net Interfaces

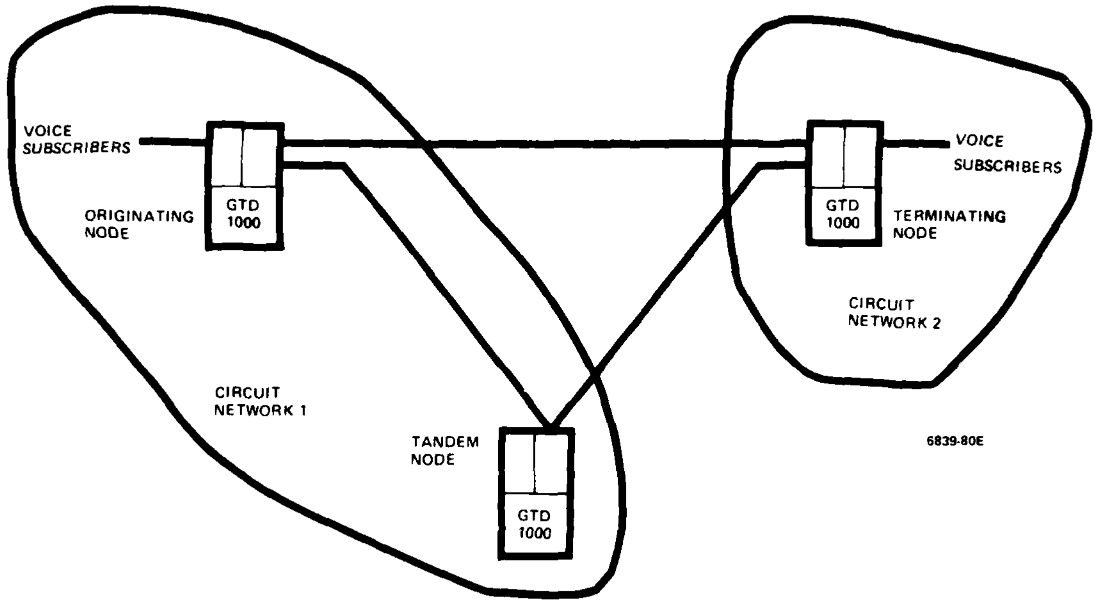
7.2.3 Experiment #2 Alternate Routing

System Description

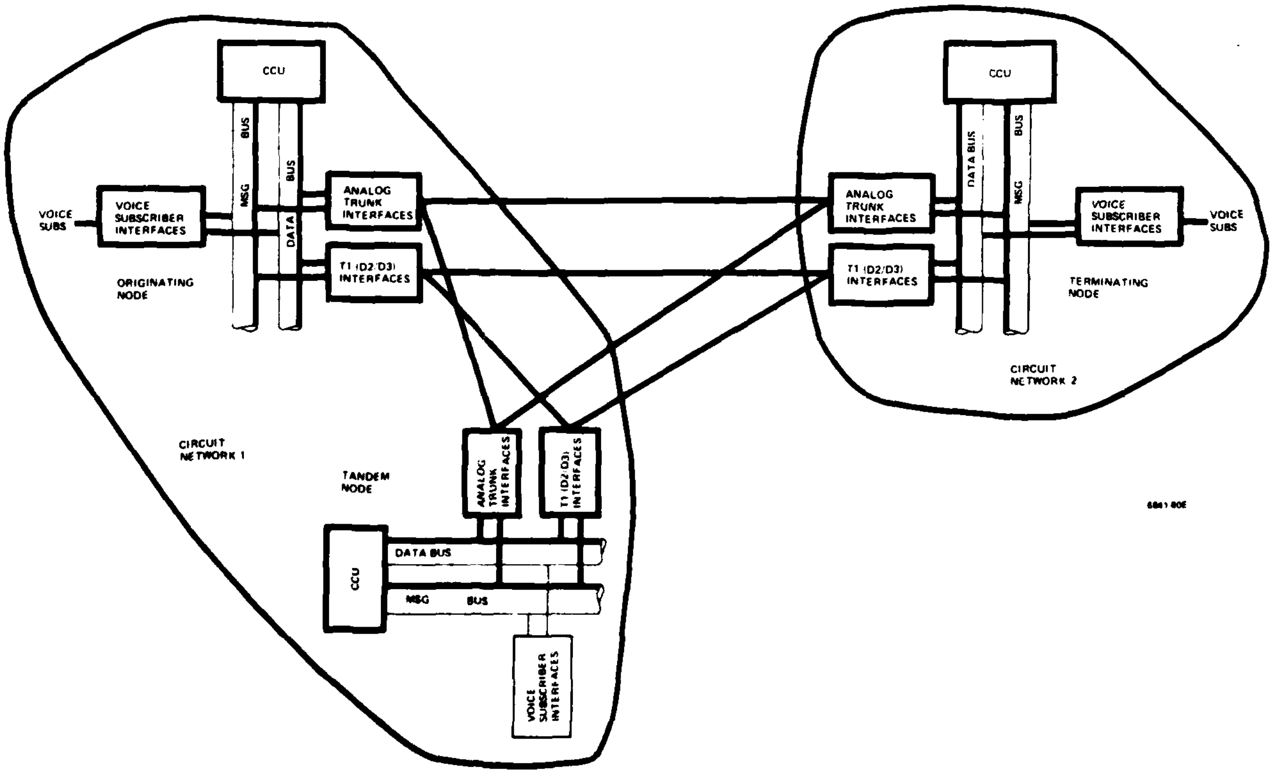
This experiment requires the addition of a third circuit switching node. There is a satellite interface required and the routing, signaling sequences and number translation will be affected. The test bed will be upgraded to improve statistic-collection capability.

Option I	Option II	Option III
Hardware Tasks:		
GTD 1000 (1PEC, 1CEC) Satellite Trunk	New Node Satellite Trunk	(Same as I)
Software Tasks:		
Alternate Routing Satellite Interface CPG Modifications Satellite Restrictions	Alternate Routing Satellite Interface T1 Interface Satellite Restrictions	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	5	104	48	1.9	7-2a
II	7	76	60	3.7	7-2b
III	5	104	48	1.9	7-2a



(a) Option 1 & 3



(b) Option 2

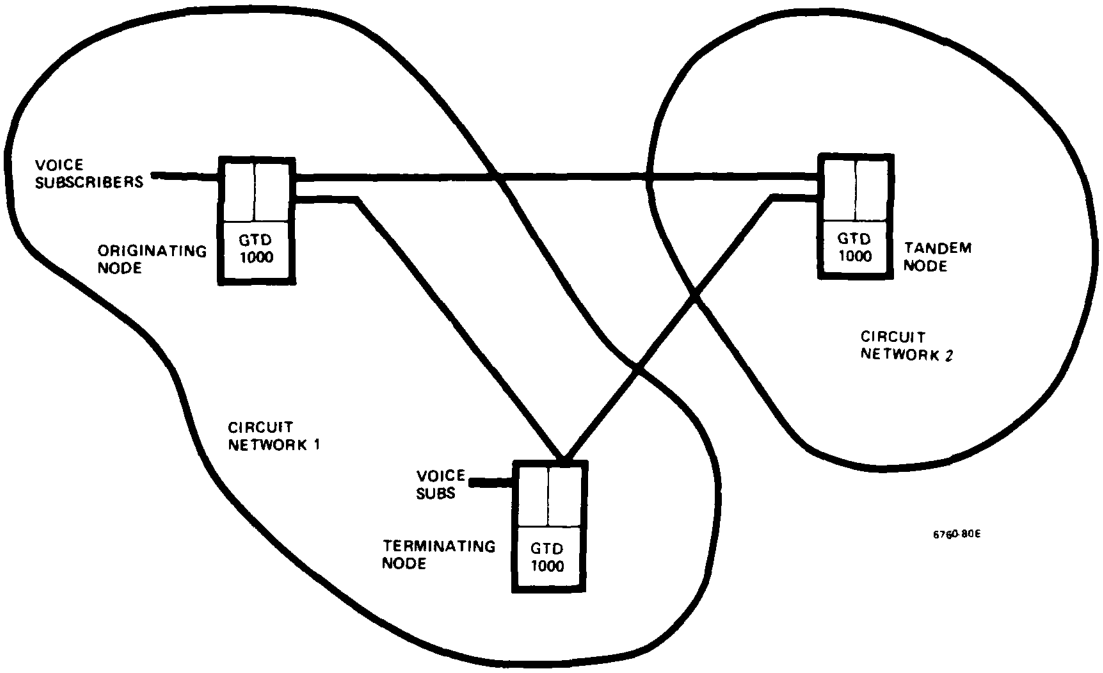
Figure 7-2. Experiment 2 - Alternate Routing

7.2.4 Experiment #3 - Tandem Routing

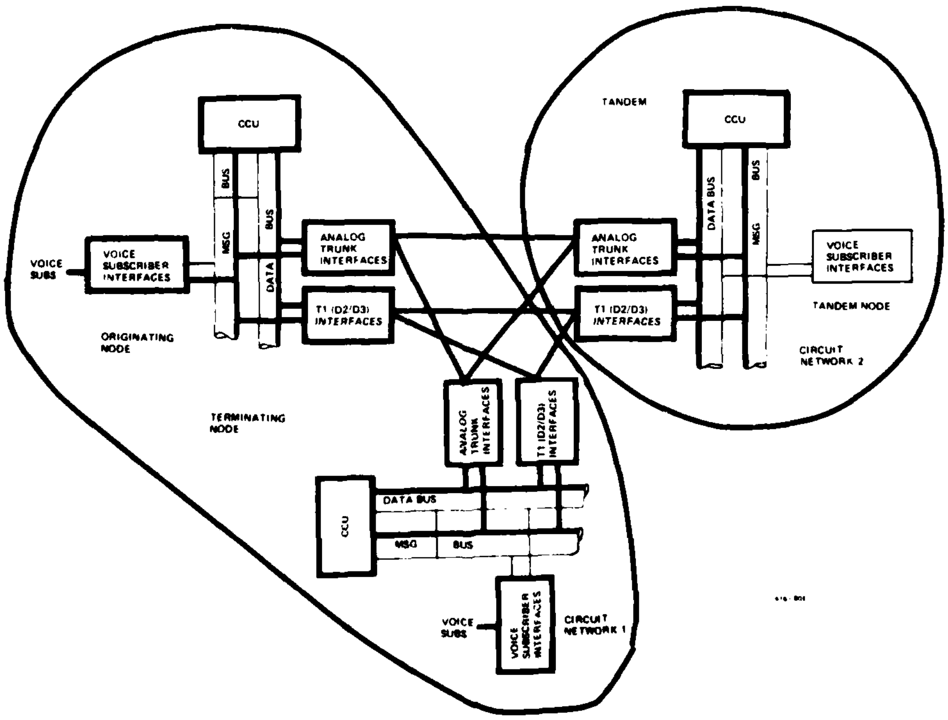
This experiment entails further refinement in software routing experiments. This set of experiments tests tandem calls through other networks, back to the same network. New two-level signaling sequences, translation, and routing algorithms are required.

Option I	Option II	Option III
Hardware Tasks:		
No Effort	No Effort	No Efforts
Software Tasks:		
Tandem Routing Internetting CPG Modifications	Internetting Tandem Routing	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	6	25	40	2.0	7-3a
II	6	8	32	1.6	7-3b
III	6	25	40	2.0	7-3a



(a) Option 1 & 3

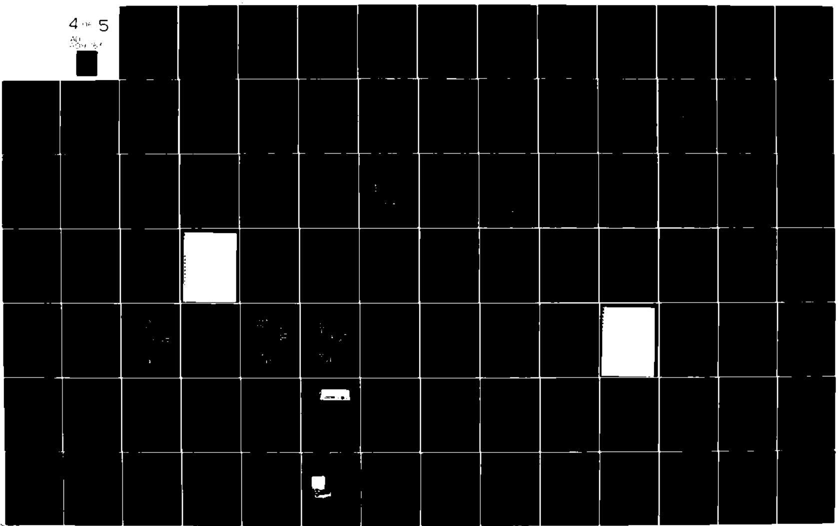


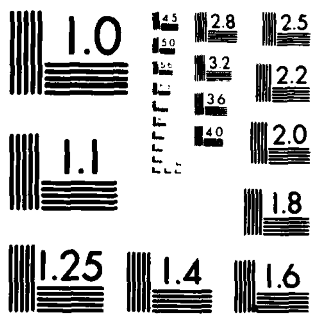
(b) Option 2

Figure 7-3. Experiment 3 - Tandem Routing

AD-A091 763 GTE PRODUCTS CORP NEEDHAM HEIGHTS MA COMMUNICATION S--ETC F/G 17/2
EXPERIMENTATION AND EVALUATION OF ADVANCED INTEGRATED SYSTEM CO--E
SEP 80 M ROSS, K GARRIGUS, J GOTTSCHALCK DCA100-79-C-0024
UNCLASSIFIED AISC/TSN-80-01 NL

4 of 5




1.0
1.1
1.25
1.4
1.6
1.8
2.0
2.2
2.5
2.8
3.2
3.6
4.0
4.5
5.0

7.2.5 Packet Baseline

System Description

This step performs a similar function as in step 1. It establishes the cost of adding packet-switch capability to the test bed including equipment, training, and software support systems. The off-the-shelf equipment of Option I and III has many more features and diagnostics than the new build of Option II. The NMC of Option I and III is also more sophisticated. Since Option II was designed specifically as a test bed, it was determined that these extras were not needed. The development level of Option II is limited to a one or two terminal type interface. It will be based on the Autovon II system but will not necessarily carry all of its features. Diagnostics and general fail-safe operation are forsaken because the goal of the test bed is to test new concepts. The baseline cost is for a single node. The costs of the additional nodes will be added in experiment #4.

Option I	Option II	Option III
Hardware Tasks:		
Telenet TP-4000 (node)	Packet Processor	(Same as I)
Telenet TP-9000 (NCC)	PAD (sync)	
Training Course	PAD (async)	
Software Development System	Packet TRK	
	NMC with x.25	
	(assumes GFE UNIX system)	
Software Tasks:		
Software Training Course:	Add Packet Swtiching	(Same as I)
System Level	Packet Trunk Interface	
TP-4000	PAD Function	
NMC		
Update/Port	CCU/NMC Interface	
Install Support Software	NMC Software	
Configuration Control		
In-House Training		

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	7	528	90	- -	7- --
II	21	151	295	29.8	7- --
III	7	528	90	- -	7- --

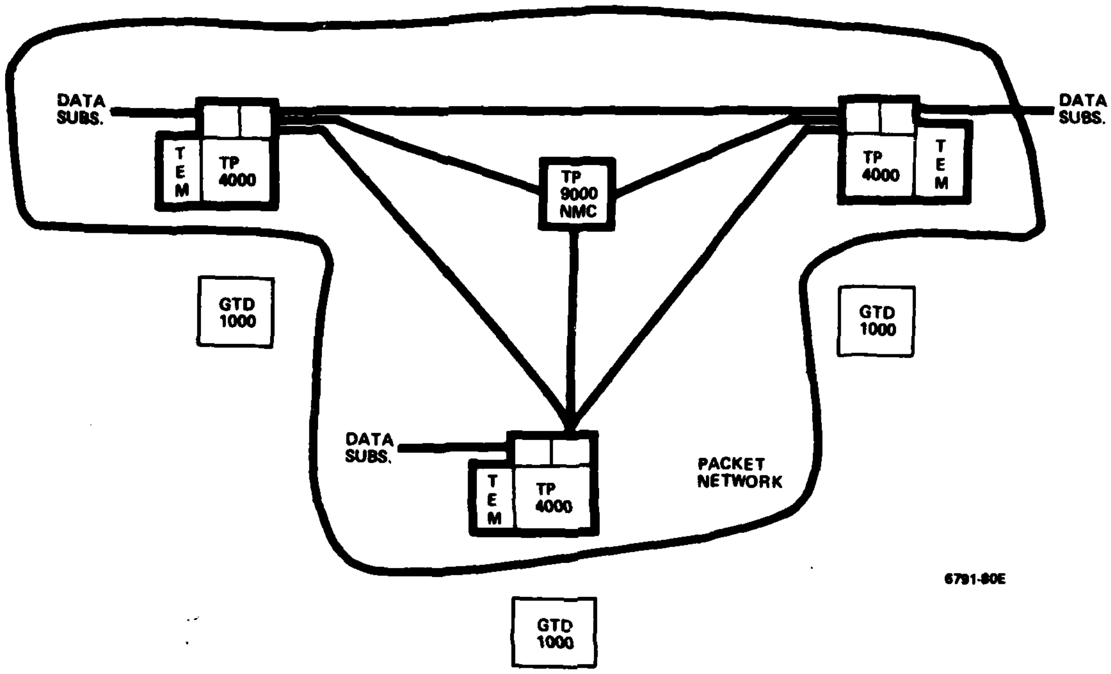
7.2.6 Experiment #4 - Packet Routing

System Description

The packet routing experiment will require different packet routing algorithms and techniques. Varieties and combinations of CRC and non-CRC packet transport are required. The trunk modules will be modified to handle the variations. Traffic Emulation capabilities to test the dynamic characteristics of the different options will be added. Data collection capabilities are required to be added to each node and to the NMC.

Option I	Option II	Option III
Hardware Tasks:		
Design TEM Module Mod for CRC Option Add 2 Nodes	Design TEM Module Mods for CRC Option Add 2 Nodes	(Same as I)
Software Tasks:		
PAR Routing PAR/PVC Routing Measurement Capability Traffic Emulation NMC Modifications	PAR Routing MIXD Routing Modify PAD Function Data Collection Traffic Emulation NMC Modifications	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	10	259	120	6.5	7-4a
II	7	50	101	7.0	7-4b
III	10	259	120	6.5	7-4a



(a) Option 1 & 3

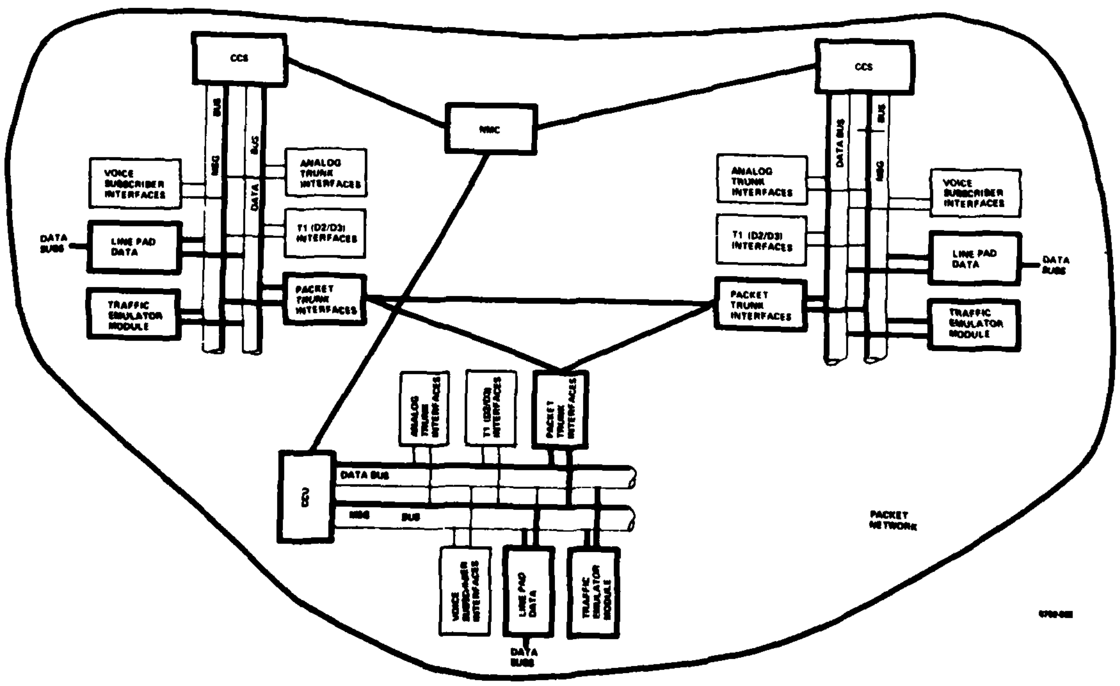


Figure 7-4. Experiment 4 - Packet Routing

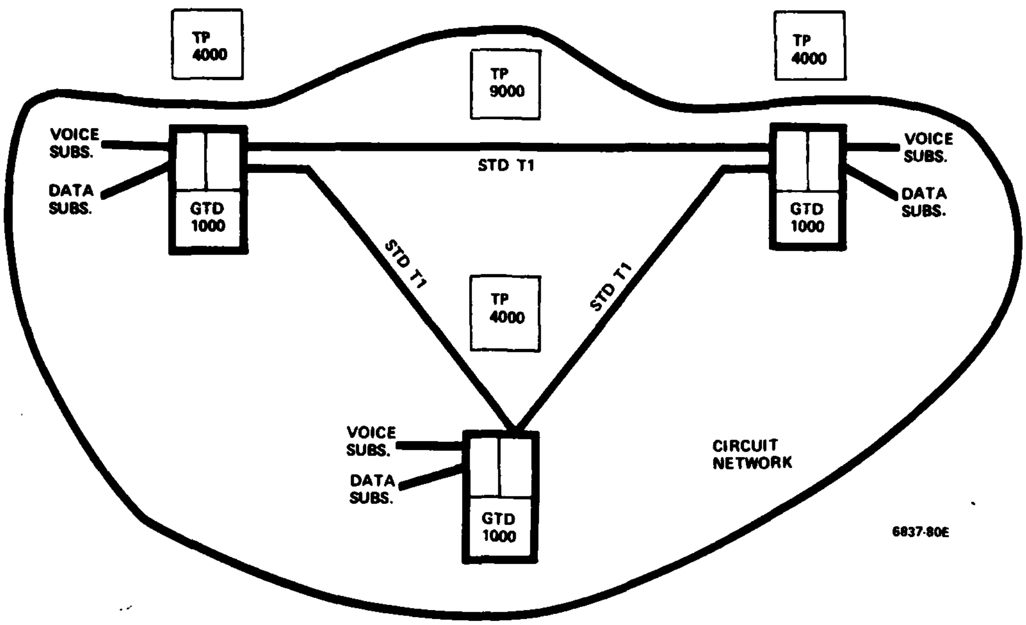
7.2.7 Experiment #5 T1 - Voice/Data

System Description

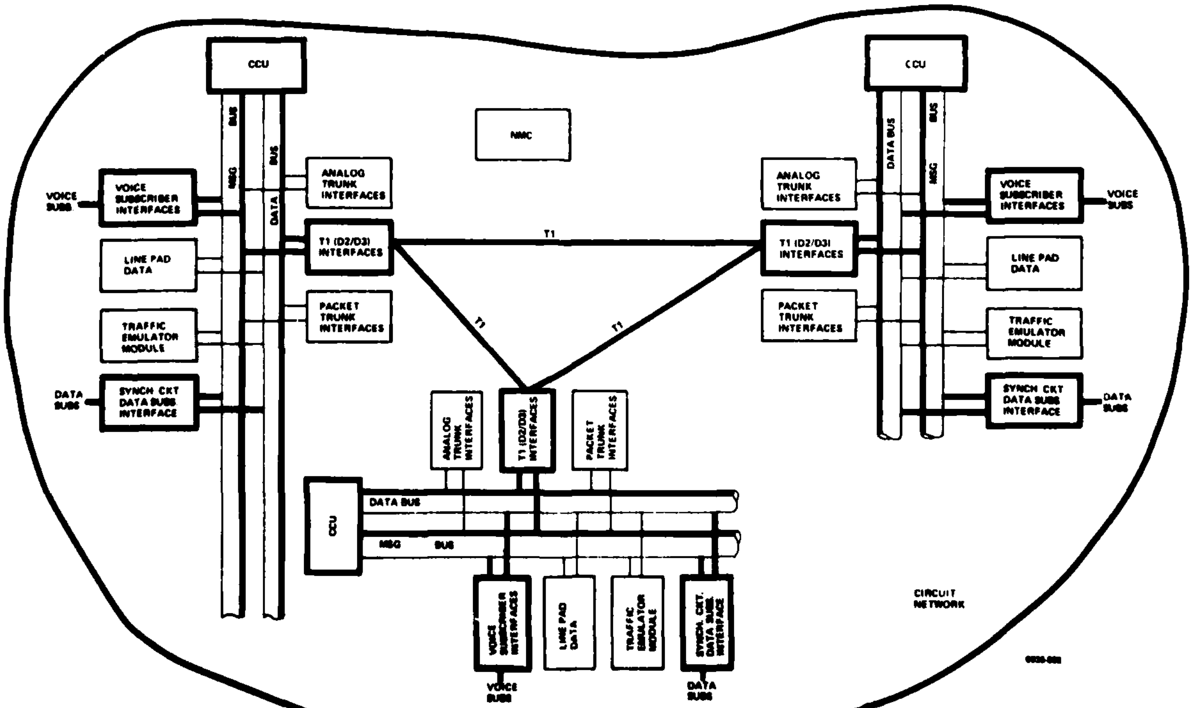
This experiment provides the circuit switch with data subscriber capability. The following will be required: 1. A data terminal circuit. 2. The associated classmark and routing restrictions to handle pure digital transmission. 3. The terminal interface for limited data rate transfer. 4. Data error rate for transmission.

Option I	Option II	Option III
Hardware Tasks:		
Digital Subscriber Card for GTD-1000	Digital Subscriber Card	(Same as I)
Error Meas. Equip.	Error Meas. Equip.	
Software Tasks:		
Data Interface Addition	Data Interface Addition	(Same as I)
Traveling Classmark	Traveling Classmark	
CPG Modifications		

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	8	29	48	0.6	7-5a
II	8	20	45	0.5	7-5b
III	8	29	48	0.6	7-5a



(a) Option 1 & 3



(b) Option 2

Figure 7-5. Experiment 5 - T1 Voice/Data

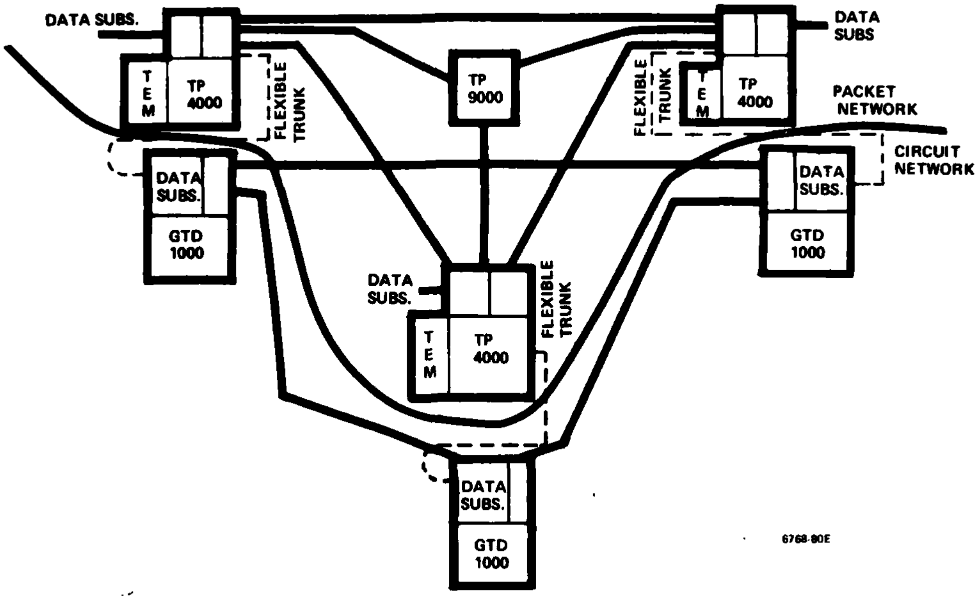
7.2.8 Experiment #6 - Flexible Trunk

System Description

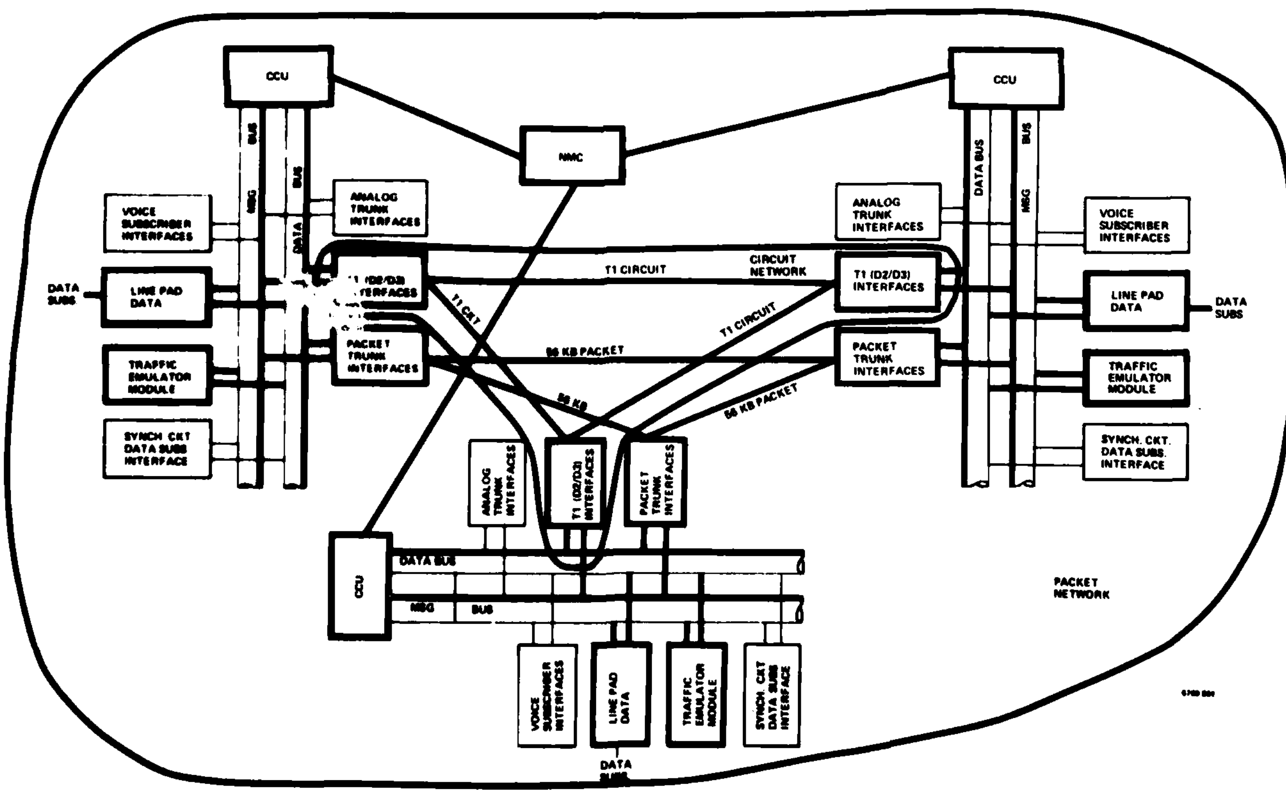
This experiment will add a circuit switch interface to the packet switch to look like a data subscriber. The packet switch will initiate 56kb trunks (calls) to other packet switches through the circuit network, based on congestion conditions. New packet routing strategy and network control are necessary. Measurement capability for net efficiency is added. MLPP interactions for voice and data are also enhanced. Options I and III will use pure external interfaces. Option II will use the internal switch matrix for direct interface and will not require an external hardware interface. The software modules will be written such that the appearance of the external interface is given.

Option I	Option II	Option III
Hardware Tasks:		
New Data Subscriber for Telenet	None (based on pure software)	(Same as I)
Software Tasks:		
Packet Switch Additions New Classmark New Routing Modify Update/Port	Routing Modifications Data Collection NMC Modification Overflow Modification	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	8	35	65	2.1	7-6a
II	6	- -	36	2.0	7-6b
III	8	35	65	2.1	7-6a



(a) Option 1 & 3



(b) Option 2

Figure 7-6. Experiment 6 - Flexible Trunks

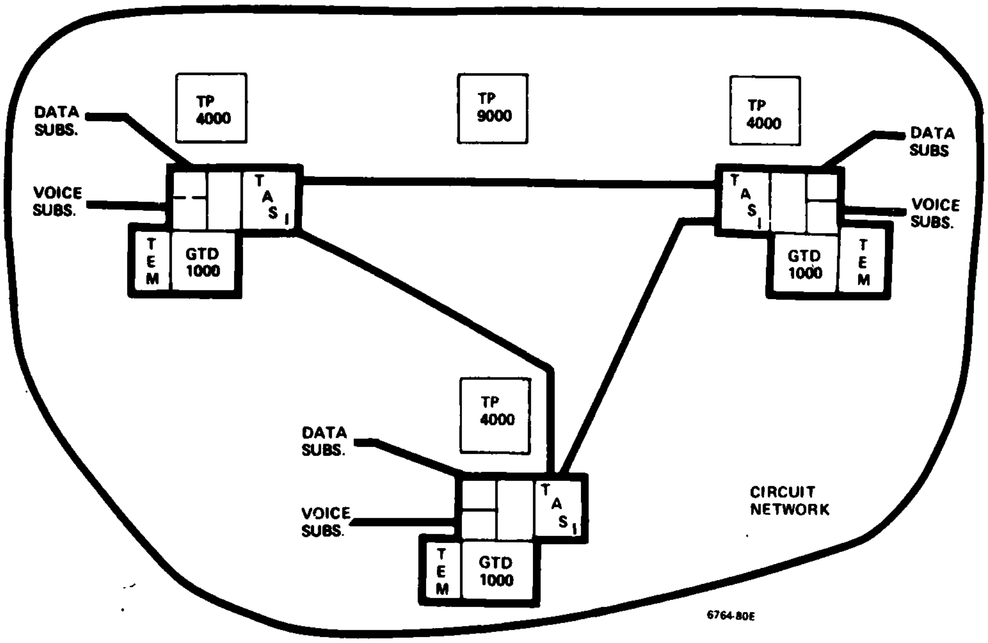
7.2.9 Experiment #7 - TASI

System Description

The TASI Experiment will provide a TASI capability based on D2/D3 format. The TASI trunk will be 24 of 48 lines. The major change to the trunks will be the addition of voice silence detectors. The GTD-1000 trunk card will be a totally new modification where two trunks will be combined at the outlet. The option two version will be a modification and addition to the T1 trunk developed for the baseline switch.

Option I	Option II	Option III
Hardware Task:		
Silence Detectors Modify GTD-LPU for TASI TRK Combine two T1s into 1 TASI	Silence Detector TASI Control on T1 TRK	(Same as I)
Software Tasks:		
None	None	None

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	10	46	82	- -	7-7a
II	10	18	67	- -	7-7b
III	10	46	82	- -	7-7a



(a) Option 1 & 3

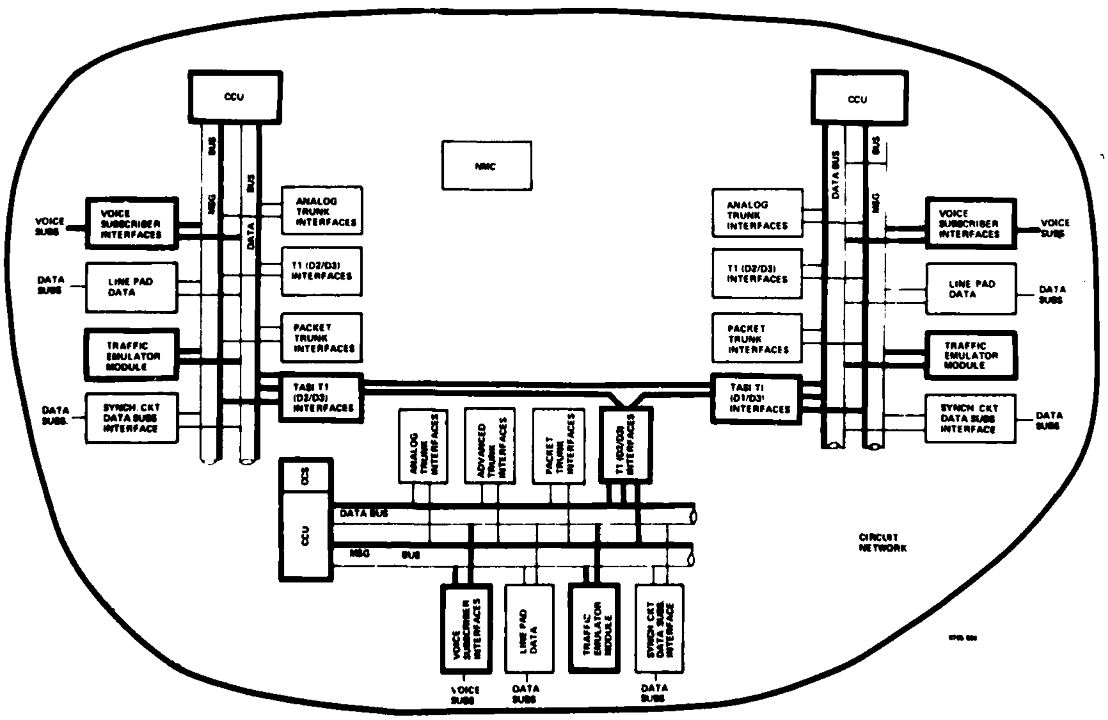


Figure 7-7. Experiment 7 - TASI

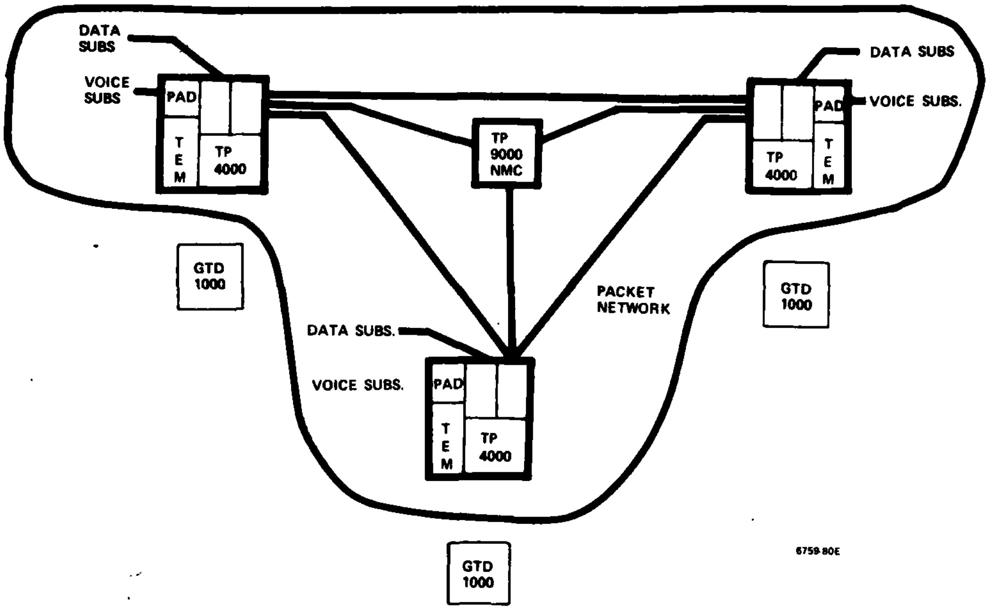
7.2.10 Experiment #8 - Packetized Voice

System Description

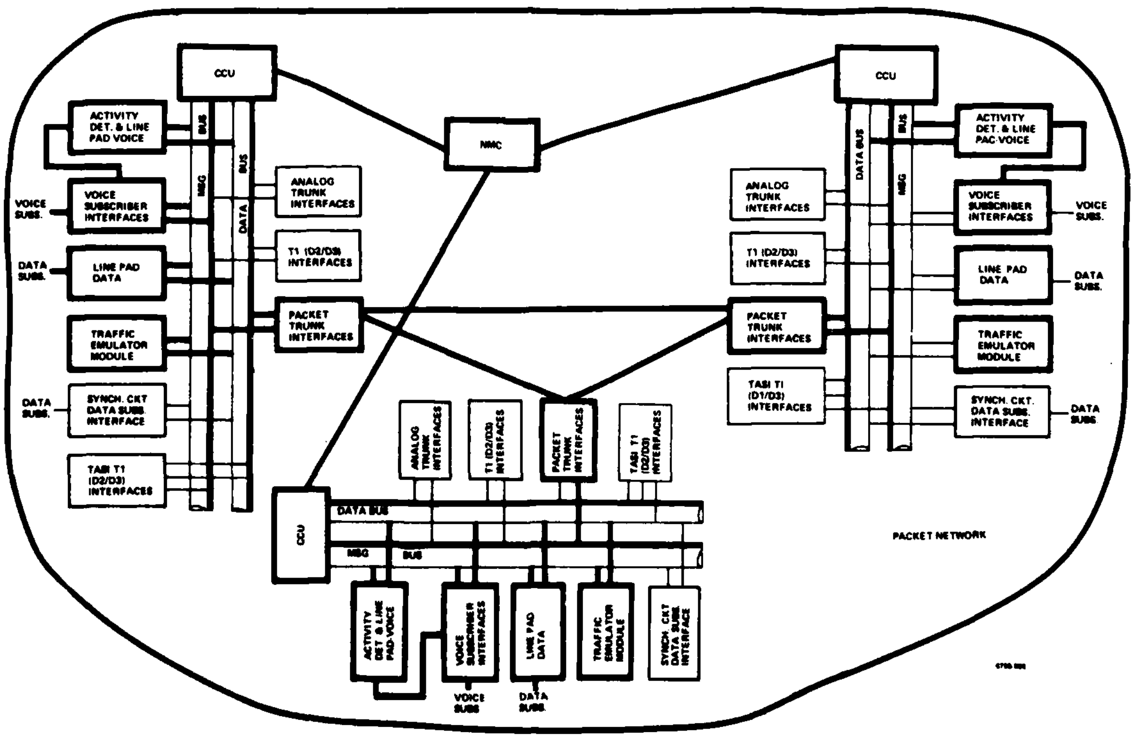
This experiment will use the silence detector design of Experiment 7 to build a new voice PAD function. Delay measurements and TEM enhancements will be added to accomodate the packet voice.

Option I	Option II	Option III
Hardware Tasks:		
Voice PAD (Telenet)	Voice PAD	(Same as I)
Software Tasks:		
Voice Interface Addition to Packet Switch	PAD Function for Voice	(Same as I)
PAD Function for Voice	Voice Classmarks	
Update/Port Modes		

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	6	19	44	0.9	7-8a
II	6	7	40	0.8	7-8b
III	6	19	44	0.9	7-8a



(a) Option 1 & 3



(b) Option 2

Figure 7-8. Experiment 8 - Packetized Voice

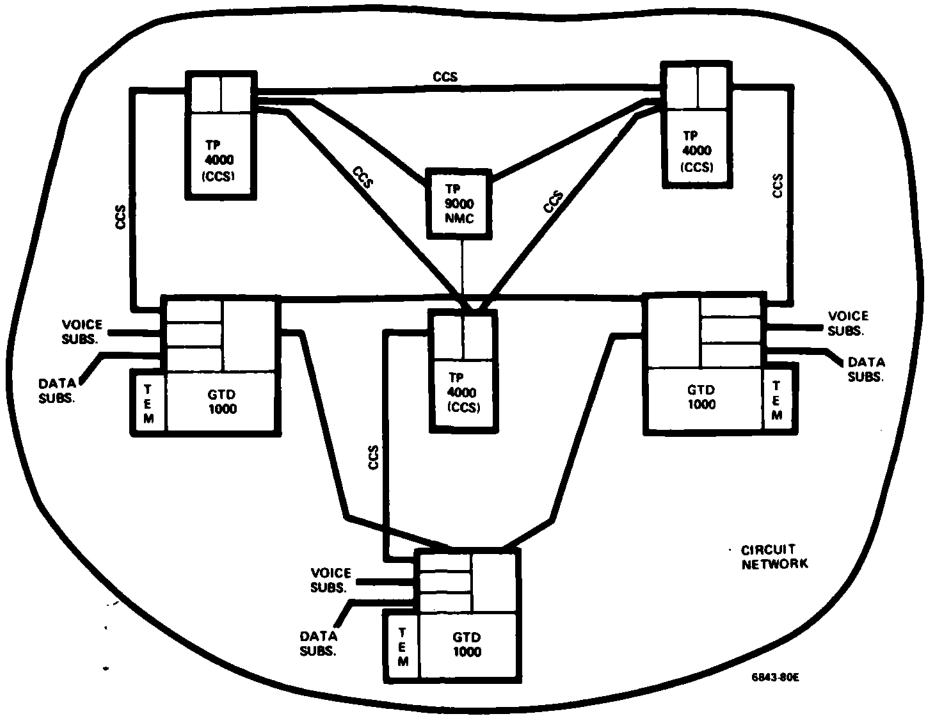
7.2.11 Experiment #9 - CCS

System Description

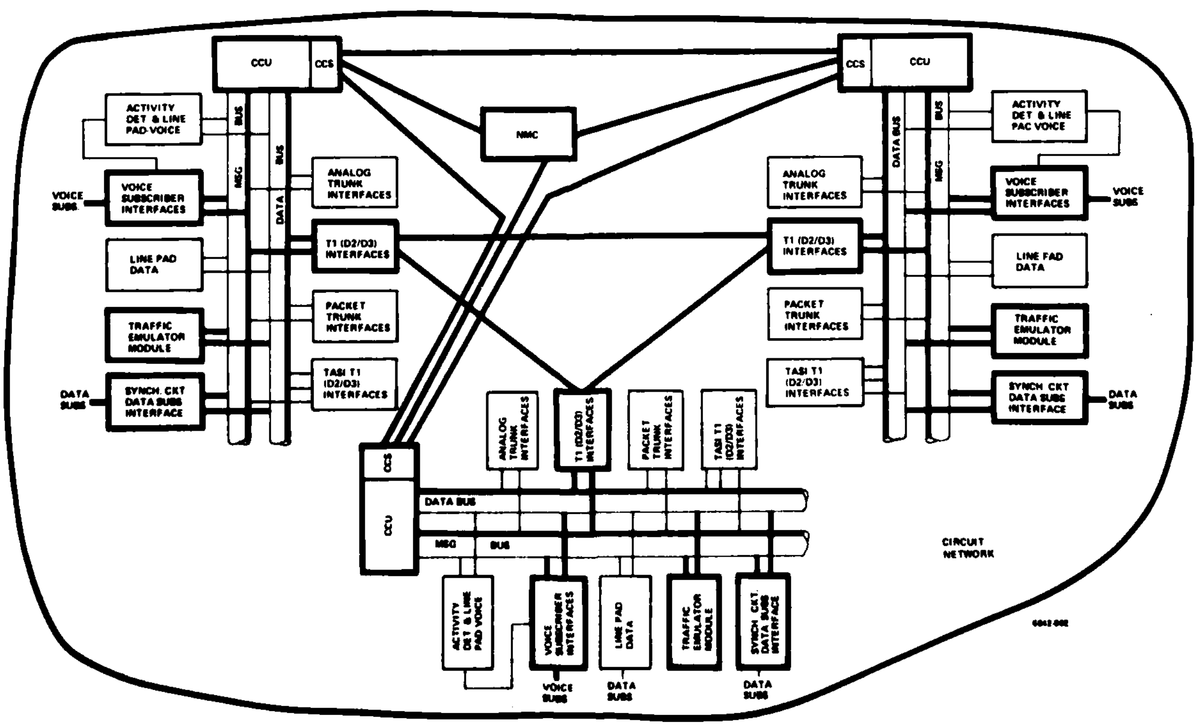
The CCS experiments assume a CCITT #7 format. This will require the addition of new packetizing format control. The experiment is to incorporate #7 capability for the previously tested features. The approach for the two options is different. The GTD version will use a standard SLI interface between the circuit switch CCU and the Telenet pad. There will be a #7 packet trunk designed for the Telenet system that will be used to interconnect the nodes. The Option II build up will develop a #7 interface that ties directly to the bus systems. Experiments will add data collection capability and distributed network control.

Option I	Option II	Option III
Hardware Tasks:		
SLI Link CCU to PAD	#7 TRK Dev.	(Same as I)
Software Tasks:		
CCITT #7 Addition	CCITT #7 Addition	(Same as I)
CPG Modifications	Classmark Additions	
Update/Port Mods	Modify x.25 Protocol	
Modify #7 for Features:	Modify #7 for Features:	
MLPP	MLPP	
Originating Office Control	Originating Office Control	
Recovery	Recovery	
Data Collection	Data Collection	

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	14	91	141	6.7	7-9a
II	12	11	112	6.3	7-9b
III	14	91	141	6.7	7-9a



(a) Option 1 & 3



(b) Option 2

Figure 7-9. Experiment 9 - CCS

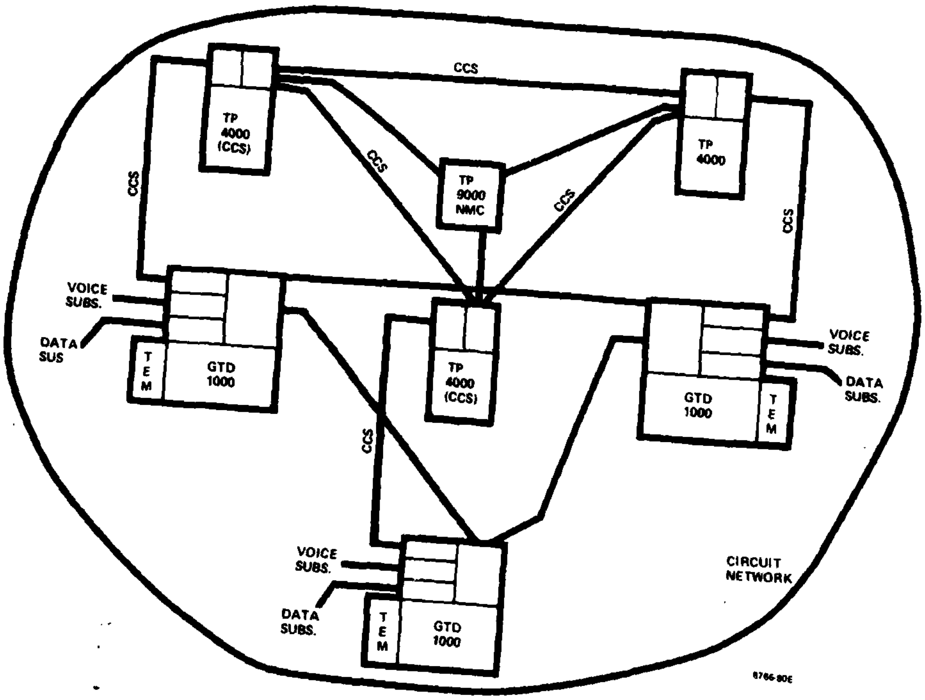
7.2.12 Experiment #10 - CCS and Features

System Description

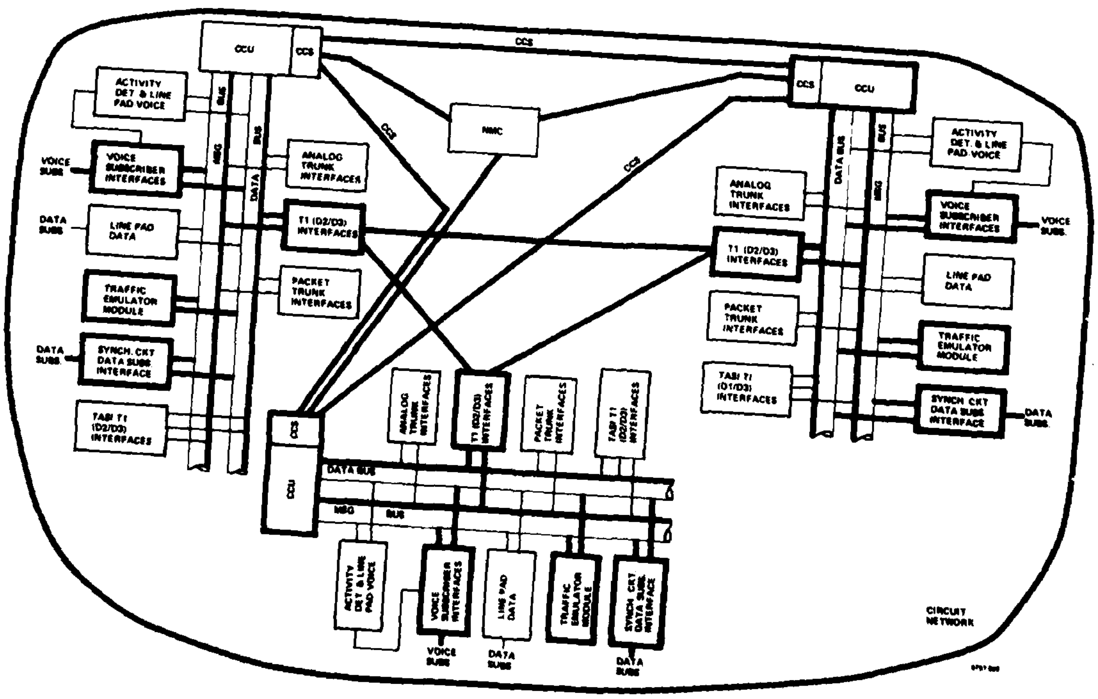
This experiment is an extension of experiment #9. Advanced features such as network call forwarding and network conferencing will be added. Centralized network data base and central network control will require the addition of #7 capability to the NMC facility.

Option I	Option II	Option III
Hardware Tasks:		
Add #7 Capability to TP-9000	Add #7 Format to NMC (DEC)	(Same as I)
Software Tasks:		
Addition of CCS to NMC	CCS Expansion	(Same as I)
Feature Additions:	Addition of CCS to NMC	
Multi-homed	Feature Additions:	
Network Call Fwd.	Multi-homed	
Mobile Subscribers	Network Call Fwd.	
Distributed Conferencing	Mobile Subscribers	
Multi-addressing	Distributed Conferencing	
	Multi-addressing	

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	14	87	129	6.8	7-10a
II	13	7	120	8.6	7-10b
III	14	87	129	6.8	7-10a



(a) Option 1 & 3



(b) Option 2

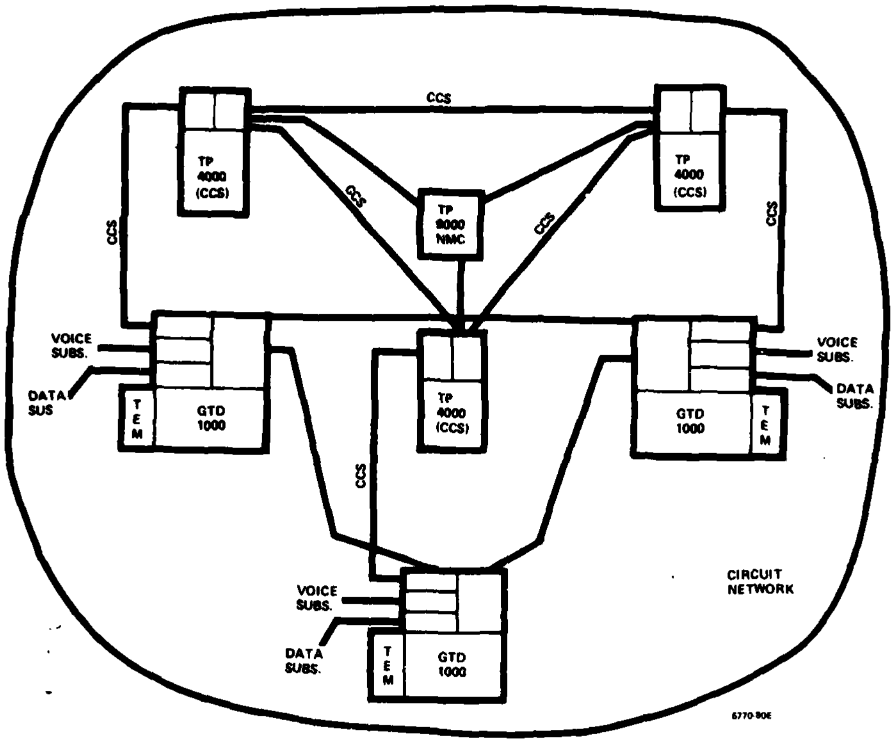
Figure 7-10. Experiment 10 - CCS/Advanced Features

7.2.13 Experiment #11 - Integrated Control System Experiment

This experiment is geared to the problems of technical control. The addition of automatic test equipment and interswitch control by CCS will be tested to enhance technical control. New classmarks for special calls, new CCITT #7 messages, and test equipment control processing will be added.

Option I	Option II	Option III
Hardware Tasks:		
Automatic Test Equipment	Automatic Test Equip.	(Same as I)
Software Tasks:		
Diagnostics:	Control Function	(Same as I)
Automatic Tests	Additions	
Call Set Up	AMA to NMC	
Pre-Determined Routing	Diagnostics:	
Tracing	Automatic Tests	
Measurement Additions	Set up Calls	
NMC Modifications	Pre-Determined Routing	
CPG Modification	Tracing	

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	11	140	94	4.4	7-11a
II	8	66	175	4.8	7-11b
III	11	140	94	4.4	7-11a



(a) Option 1 & 3

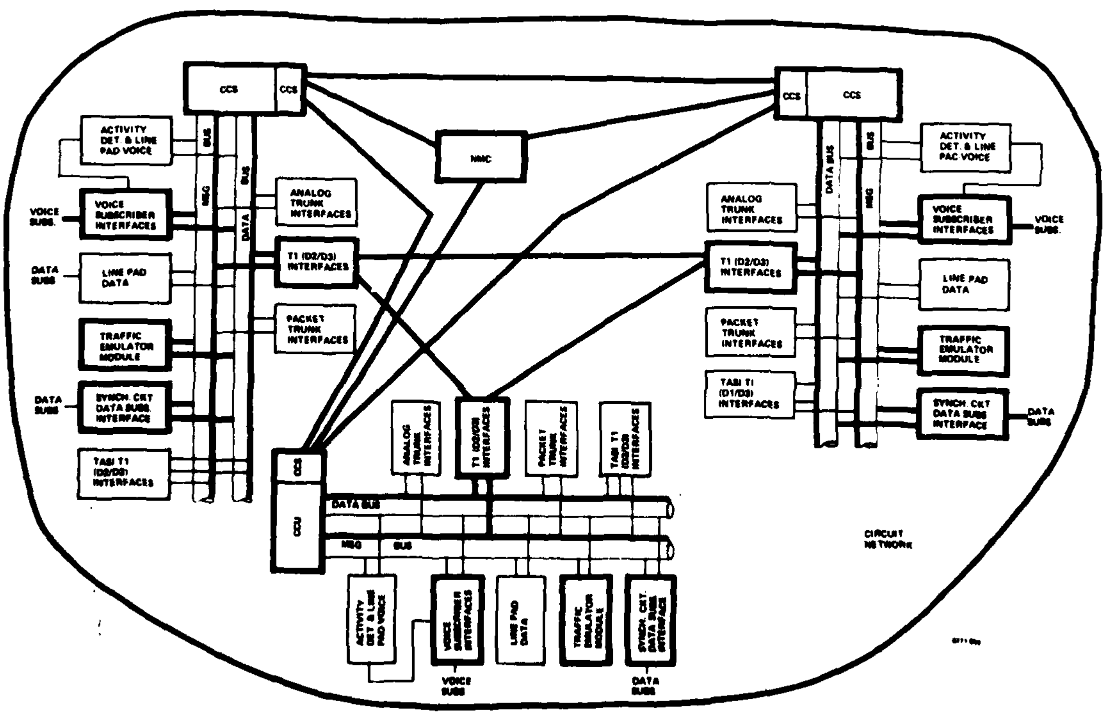


Figure 7-11. Experiment 11 - Integrated Control

7.2.14 Hybrid Base

System Description

The purpose of this step is to break out the cost of creating a true hybrid test bed. The definition of a hybrid switch is a full interaction of circuit and packet traffic on a trunk. A common matrix will be used and a hybrid processor will make the contention decisions. Option I will incorporate a fixed bandwidth matrix, namely the GTD-1000. The Telenet equipment will be interfaced through a special intelligent adapter that cross connects the two machines such that the TP-4000 thinks it is talking to a 1.5 mB trunk and the GTD thinks that it is controlling an LPU. The second option will not require any changes since it was desinged for the application. The third option is a cross between Options I and II. The GTD-1000 and TP 4000 will have adapters that interface to the hybrid bus of Option II. The basic components of Option II will have to be added to create a combined hierarchical switch. The Telenet and GTD switches will act as access functions to the hybrid node switch.

Option I	Option II	Option III
Hardware Tasks:		
TP-4000 to GTD-1000 Adapter	None	TP-4000 to Hybrid Adapter GTD-1000 to Hybrid Adapter Matrix Bus Message Bus Main Processor Soft. Dev. System New CCITT #2 Inter.
Software Tasks:		
Hybrid Processor TP-4000 and GTD-1000 Interface to Hybrid Processor CPG/Update Mods	None	New Nodal Processor CCS, Routing, Exec, TEM, etc. Matrix Control Processor to Processor Interface TEM Mods

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	12	103	143	6.4	7- --
II	- -	- -	- -	- -	7- --
III	15	79	224	16.1	7- --

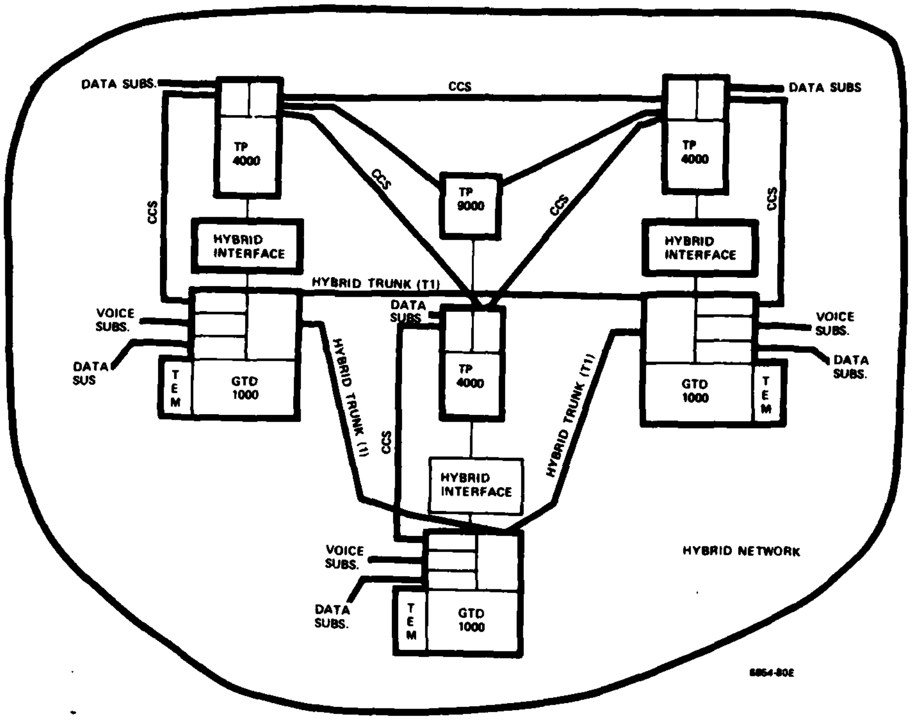
7.2.15 Experiment #12 - Combined Voice and Data

System Description

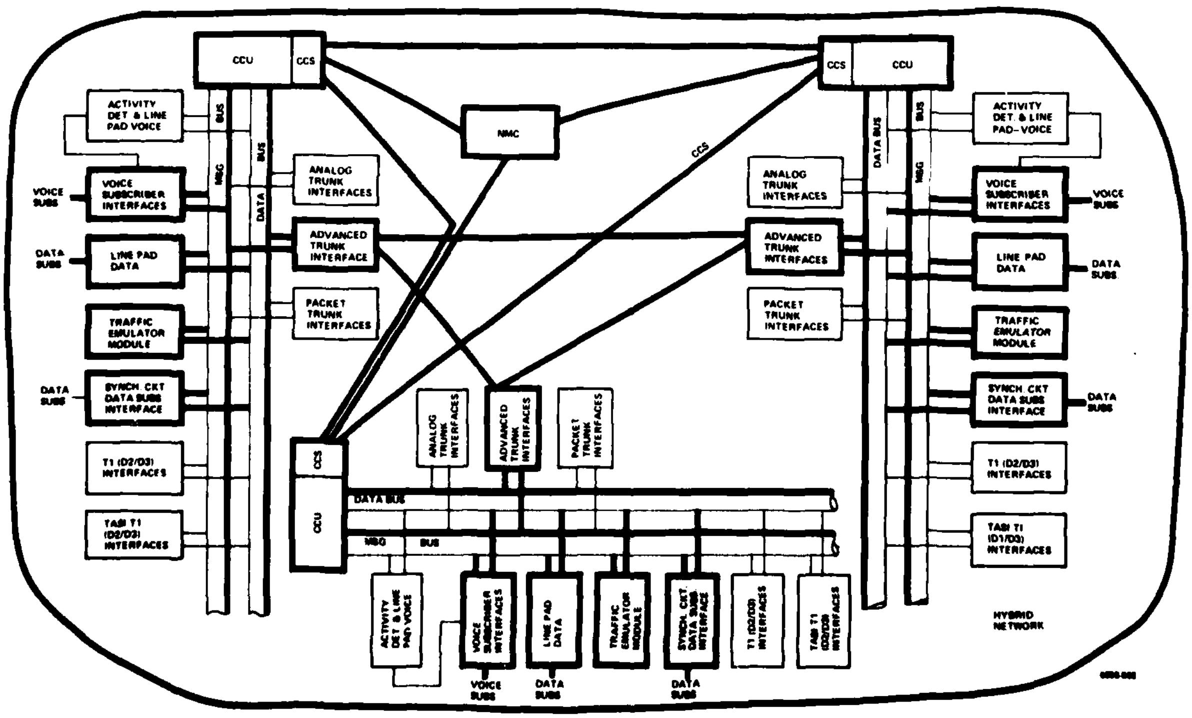
This experiment is the first true attempt at an integrated voice/data trunk. It is an extension of experiment #6 in that the packet trunk is expanded by virtue of 64KB trunk increments. The trunks are pure CCS controlled without D2/D3 (bit stealing) signalling. All options will use a common packet/circuit matrix to the hybrid trunks. The decision algorithms for bandwidth contention can be more interactive and quicker. The node is treated as a single switch for routing rather than as two separate networks as compared to the flexible trunk experiment.

Option I	Option II	Option III
Hardware Tasks:		
Modify existing T1 Trk. on GTD-1000	Modify T1 Hybrid Trk. to Remove Bit Stealing	Develop T1 Trunk Interface for Hybrid Trk.
Software Tasks:		
B/W Contention Algorithm CPT/Update Mods	B/W Contention Algorithm Database Capabilities	(Same as II)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	4	25	36	1.5	7-12a
II	3	1	22	1.1	7-12b
III	8	37	43	1.1	7-12c

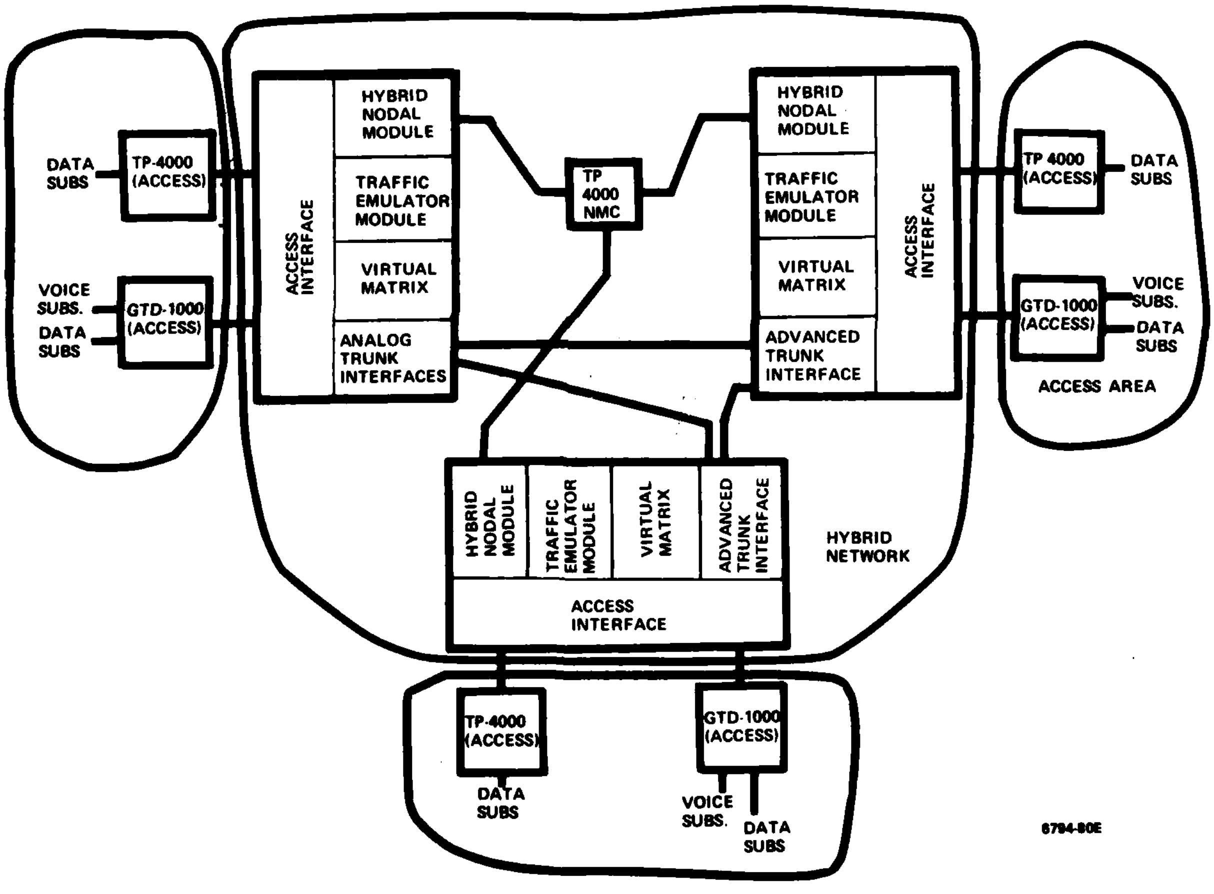


(a) Option 1



(b) Option 2

Figure 7-12. Experiment 12 - Combined Voice/Data



(c) Option 3

Figure 7-12. Experiment 12 - Combined Voice/Data

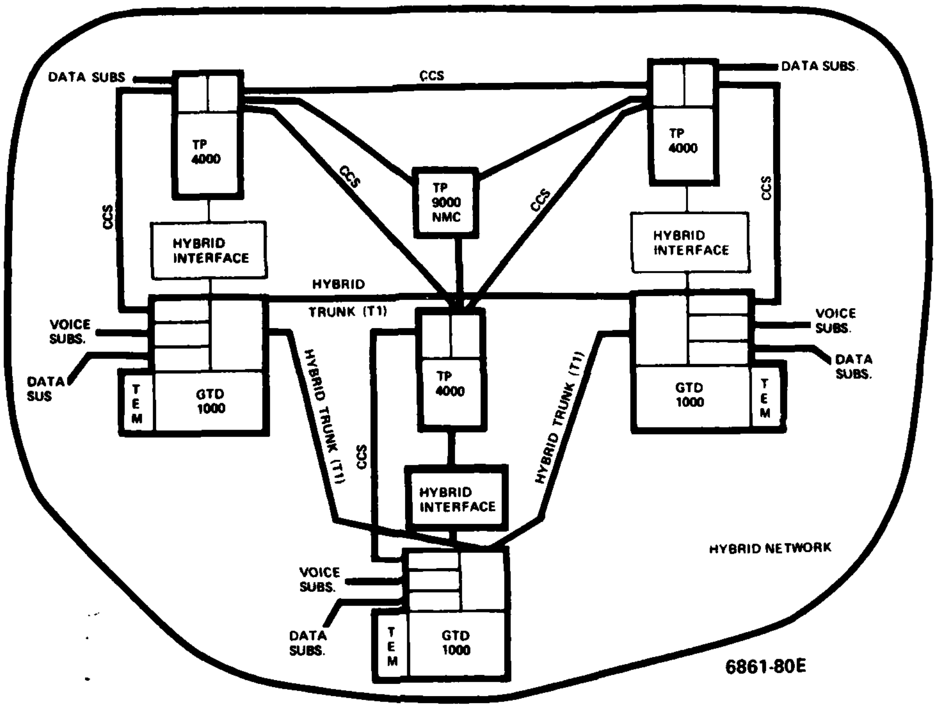
7.2.16 Experiment #13 - Network Control

System Description

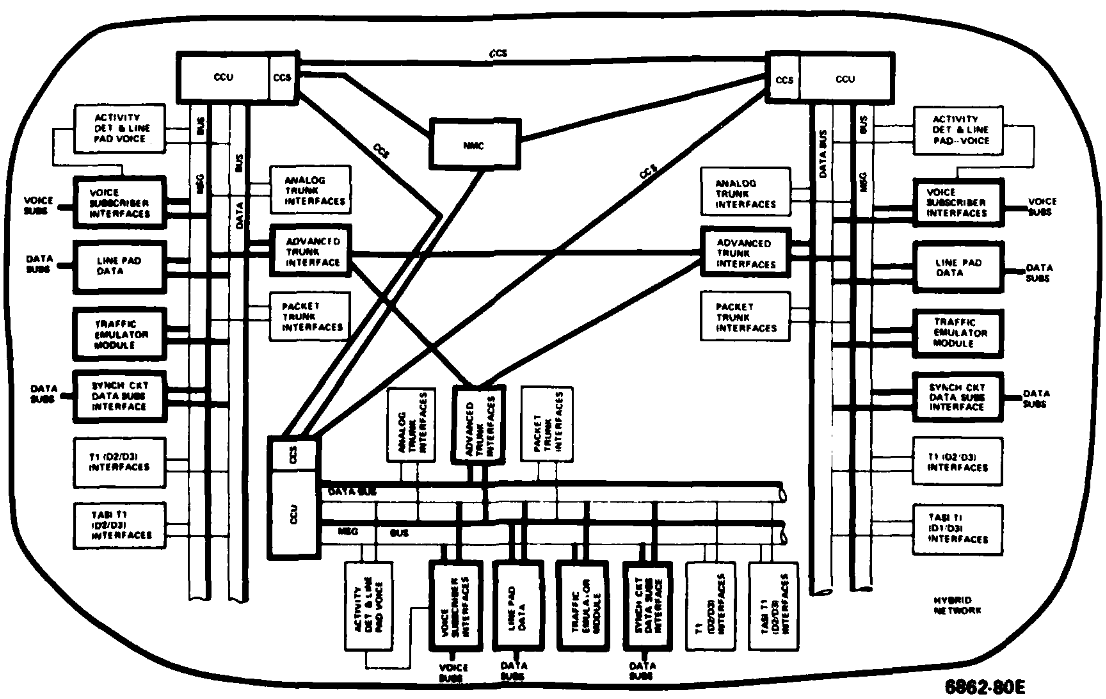
This experiment will enhance the NMC and Switch functions to explore the different characteristics of centralized hybrid control versus distributed hybrid control.

Option I	Option II	Option III
Hardware Tasks:		
None	None	None
Software Tasks:		
NMC Routing Algorithms Switch Data Base Alterations	(Same as I)	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	4	13	23	1.4	7-13a
II	4	- -	23	1.4	7-13b
III	4	13	23	1.4	7-13c

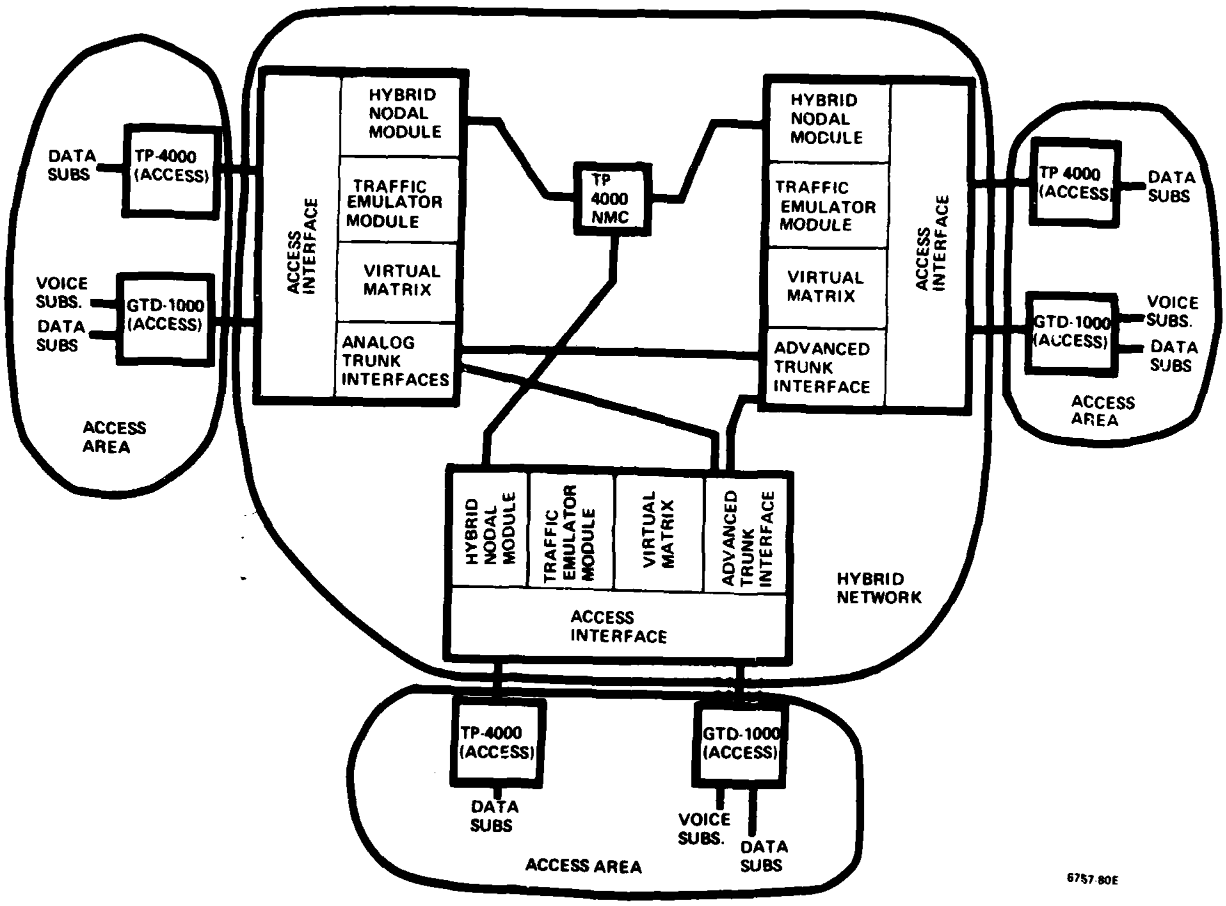


(a) Option 1



(b) Option 2

Figure 7-13. Experiment 13 - Hybrid Network Control



(c) Option 3

Figure 7-13. Experiment 13 - Hybrid Network Control

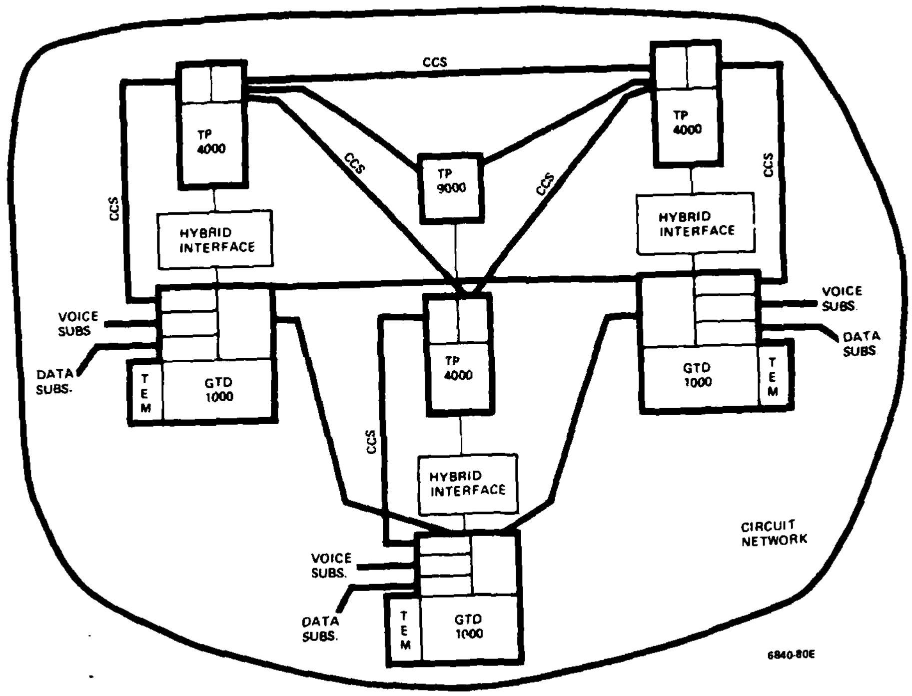
7.2.17 Experiment #14 - External Monitoring

System Description

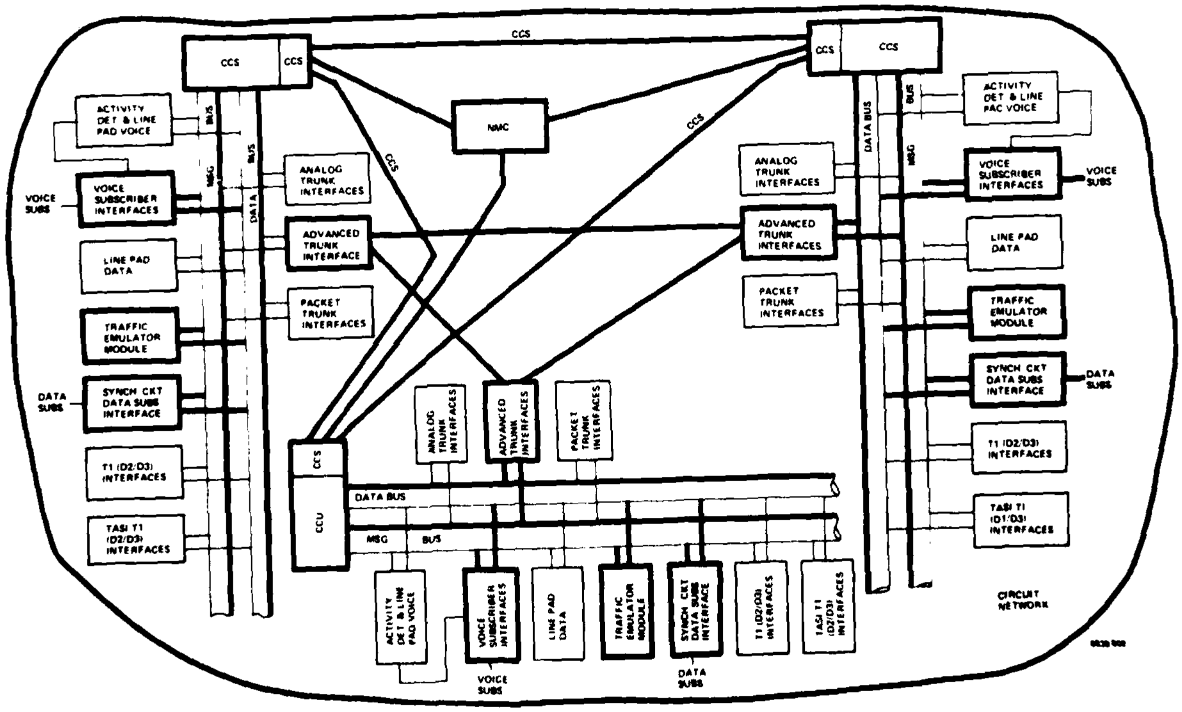
This experiment involves the addition of automatic tech control functions to older switches. Automatic test equipment will be connected to the circuit network to test the effectiveness of non-integrated testing.

Option I	Option II	Option III
Hardware Tasks:		
Automatic Test Eq.	(Same as I)	(Same as I)
Software Tasks:		
Test Box Processor NMC Modifications	(Same as I)	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	6	88	40	2.5	7-14a
II	6	66	45	2.5	7-14b
III	6	87	40	2.5	7-14c

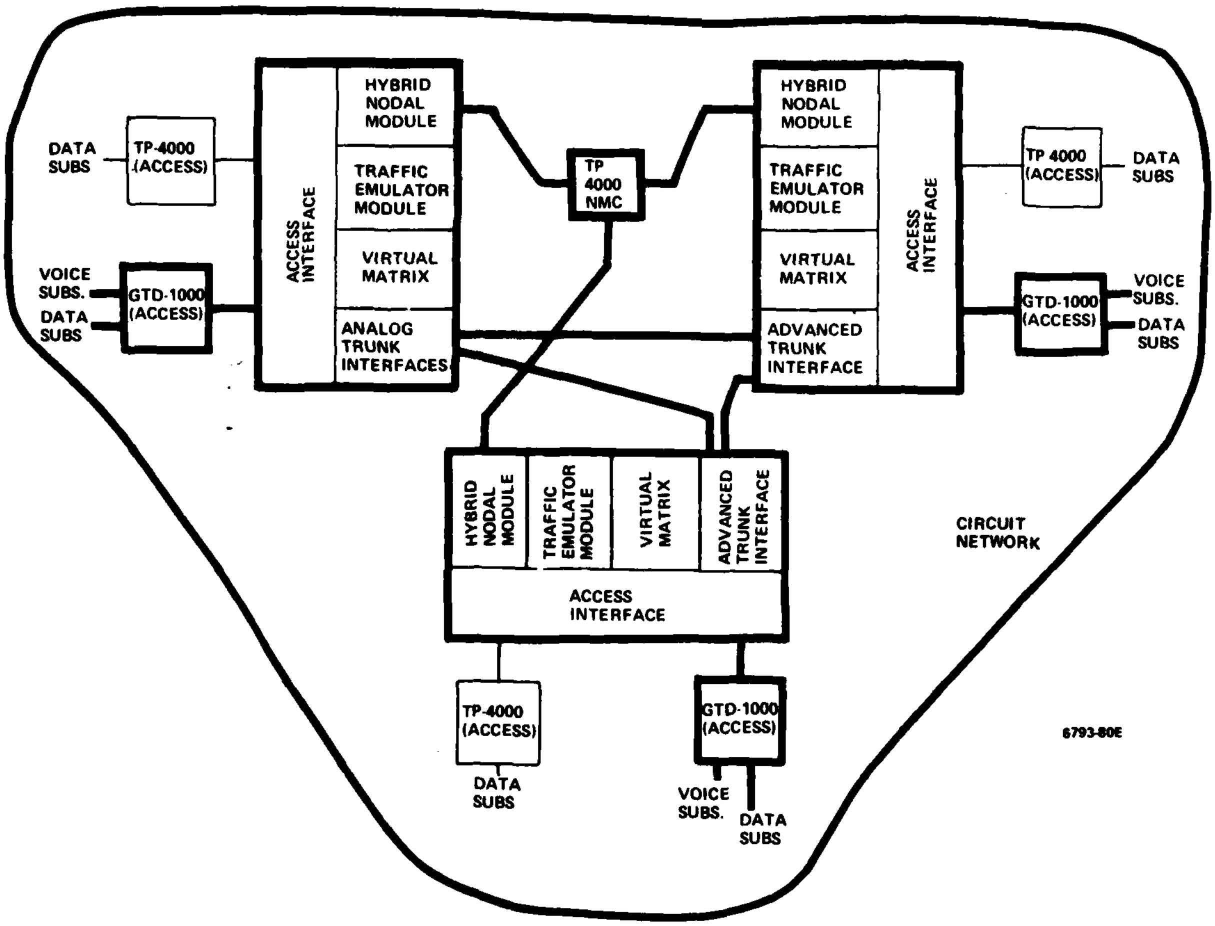


(a) Option 1



(b) Option 2

Figure 7-14. Experiment 14 - External Monitoring



(c) Option 3

Figure 7-14. Experiment 14 - External Monitoring

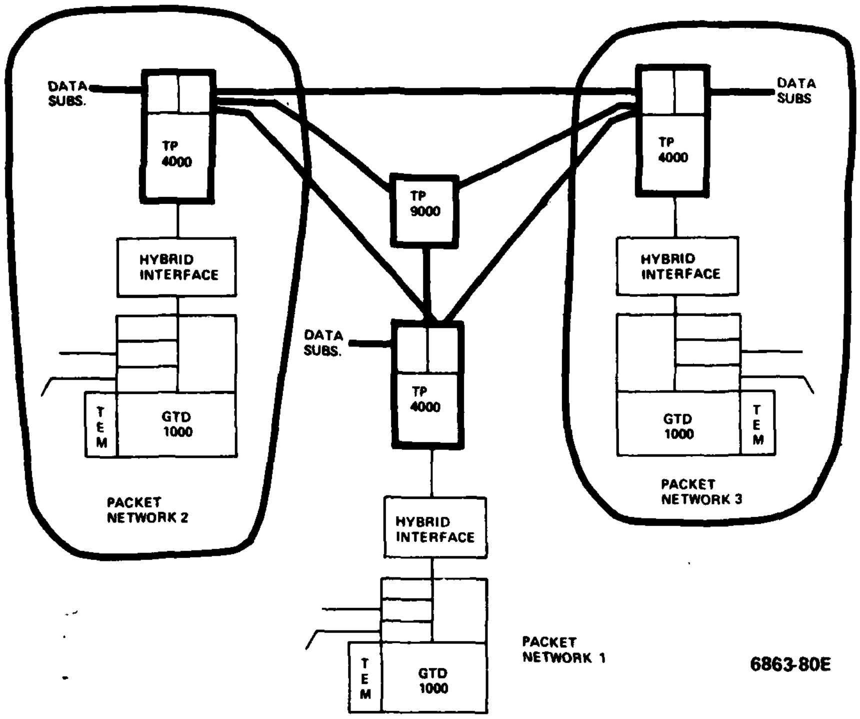
7.2.18 Experiment #15 - Data

System Description

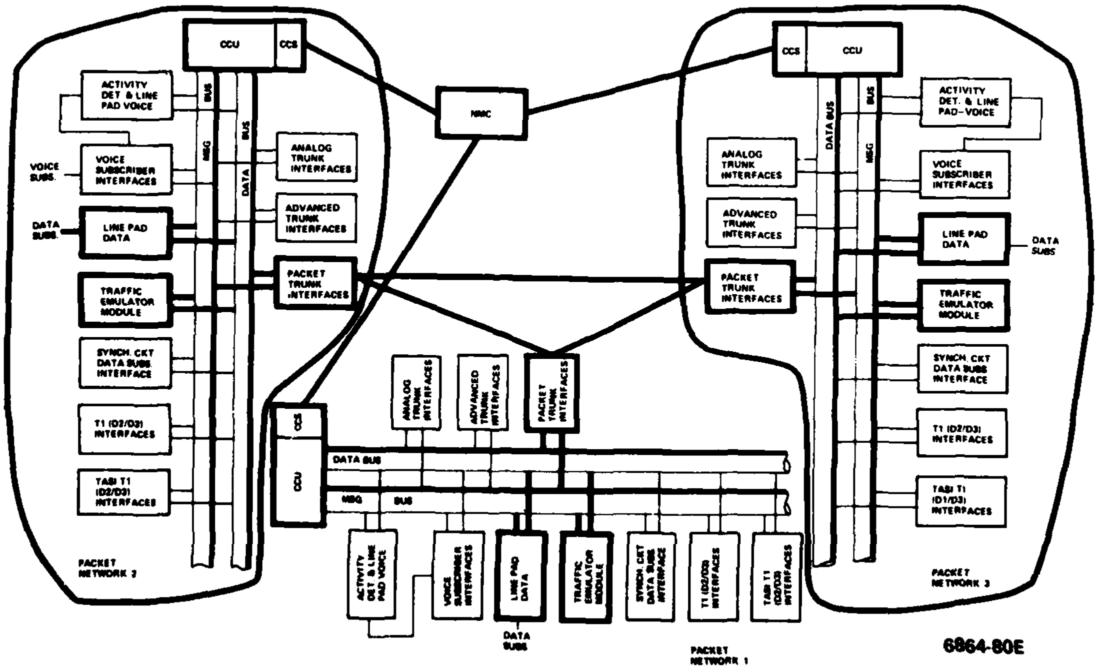
These tests are aimed at the interoperability of different packet data networks. The estimates reflect three military networks; Autodin II, WWMCCS, Arpanet. Options I and III are a more extensive effort due to a three-network design as compared to the two network design of Option II. (Option II designed as one of the three originally).

Option I	Option II	Option III
Hardware Tasks:		
Autodin II INT WWMCCS INT Arpanet INT	WMMCCS INT Arpanet INT	(Same as I)
Software Tasks:		
Additional Interfaces: WWMCCS Autodin II Arpanet Backbone Interface Conversion Port/Update Modifications	Additional Interfaces: Autodin II WWMCCS Interface Conversion	(Same as I)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	12	95	137	9.3	7-15a
II	9	8	105	7.0	7-15b
III	12	96	137	9.3	7-15c

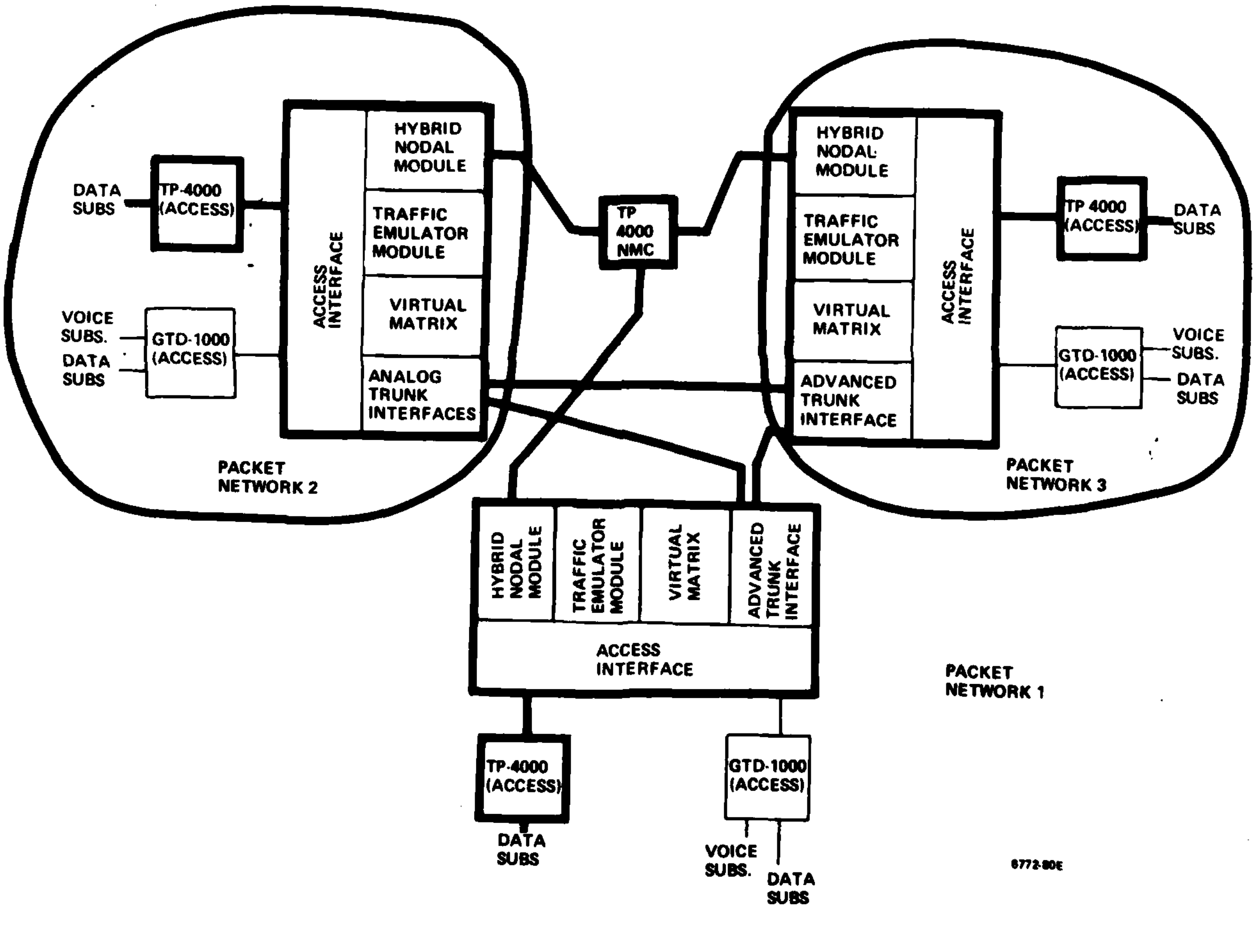


(a) Option 1



(b) Option 2

Figure 7-15. Experiment 15 - Data Experiments



(c) Option 3

Figure 7-15. Experiment 15 - Data Experiments

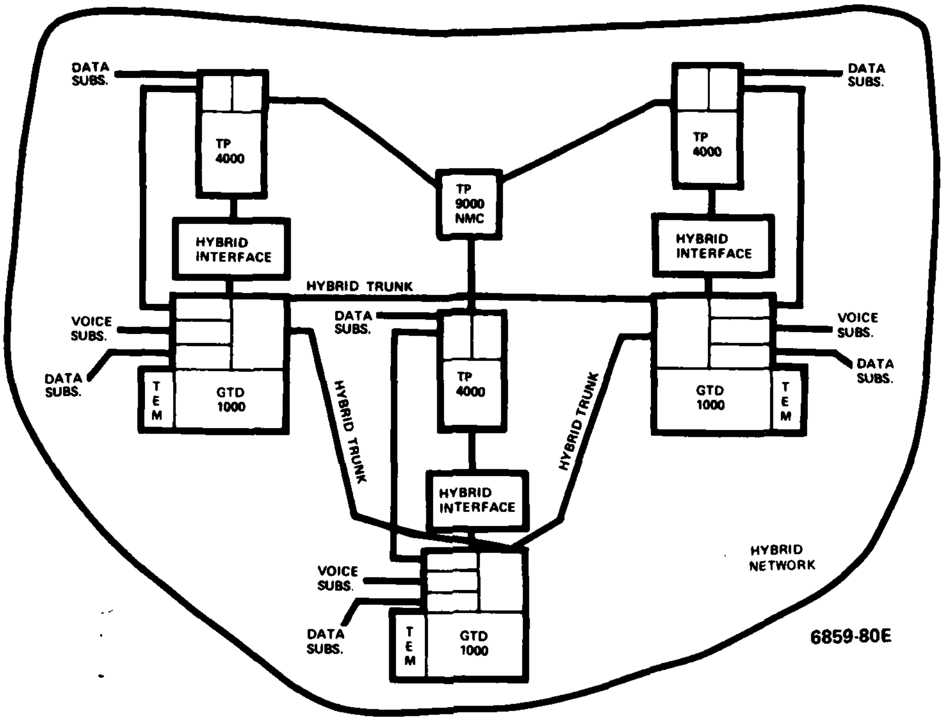
7.2.19 Experiment #16 - Variable Bound/Fixed Bandwidth

System Description

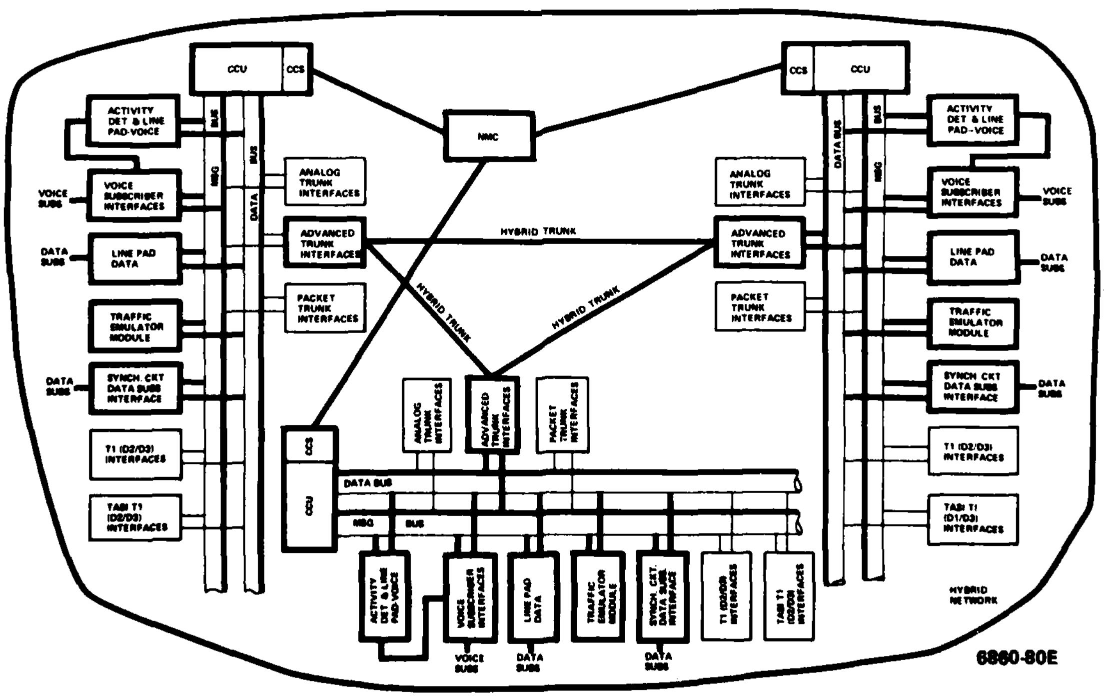
This experiment is the first to use the T1 trunk with an instantaneous packet bandwidth adjustment. The trunk can be operated with and without TASI. The trunk control and framing are integral to the trunk and not part of CCS. Each trunk is treated as a single variable bandwidth packet trunk that varies by 64KB increments. The multiplexing format for the T1 trunk still has the interleaved patterns of 8 kHz samples similar to current T1 format (Figure 4-12). Bandwidth contention algorithms, routing, and statistic gathering routines have to be adjusted.

Option I	Option II	Option III
Hardware Tasks:		
Build New T1 TRK for GTD-1000	Add Variable BW packet to T1 Trk	New T1 TRK for Hybrid Bus with TASI & Var. BW
Software Tasks:		
Enhance B/W Contention	Enhance B/W Contention Algorithms	(Same as II)
Subframe Map Control	Subframe Map Control	
Hybrid Interface Mod	Enhance Statistics	

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	9	52	82	2.8	7-16a
II	12	16	83	2.5	7-16b
III	12	46	93	2.5	7-16c

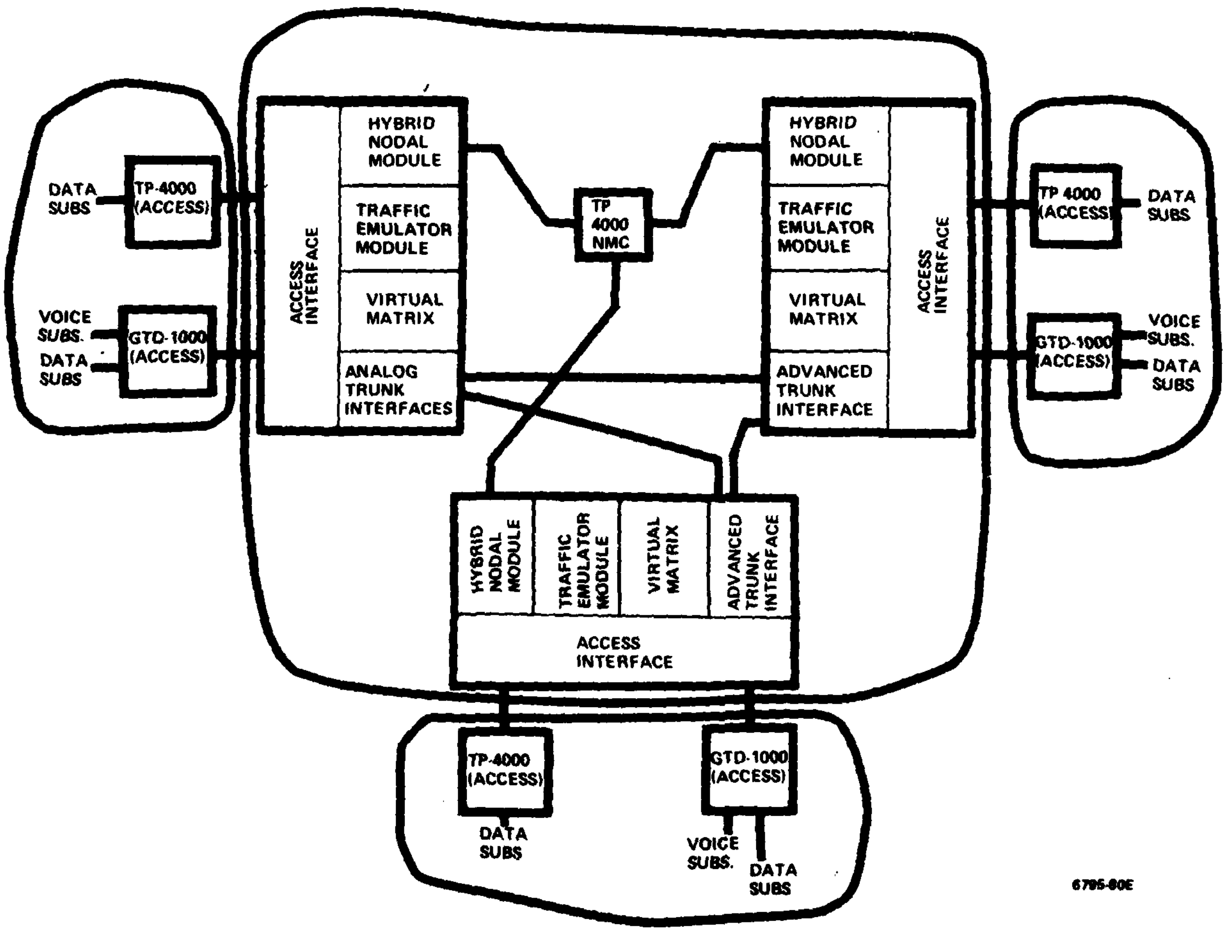


(a) Option 1



(b) Option 2

Figure 7-16. Experiment 16 - Fixed B/W, Variable Boundary



(c) Option 3

Figure 7-16. Experiment 16 - Fixed B/W, Variable Boundary

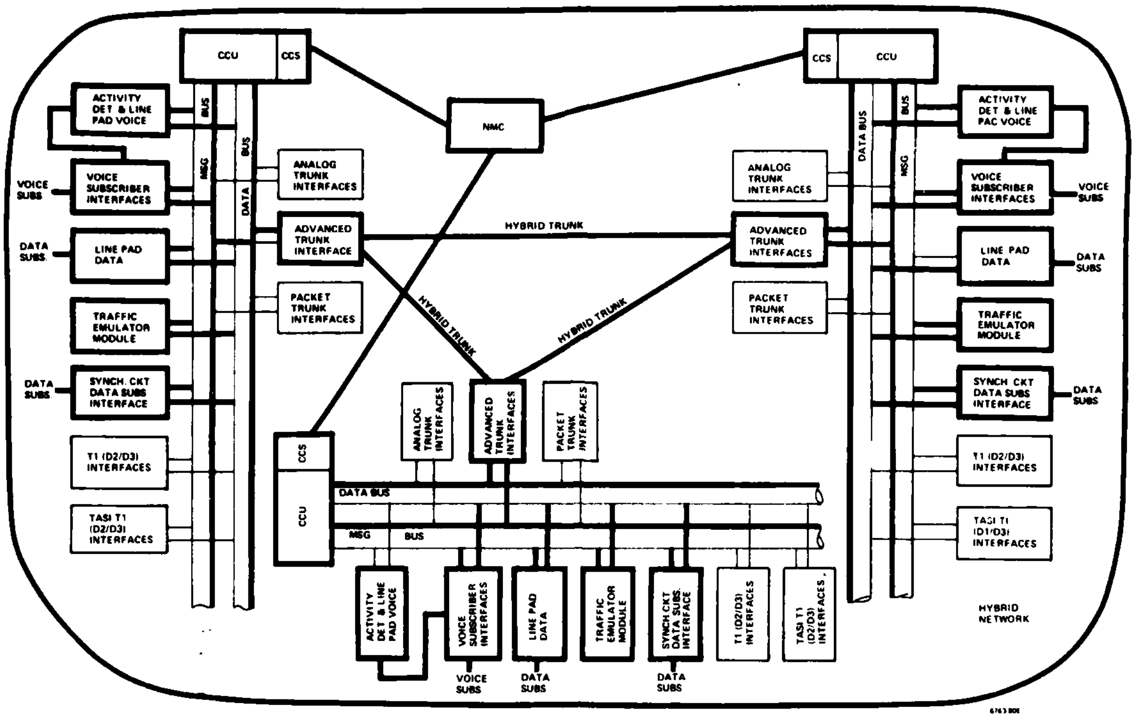
7.2.20 Experiment #17 - Variable Frame and Variable Bandwidth

System Description

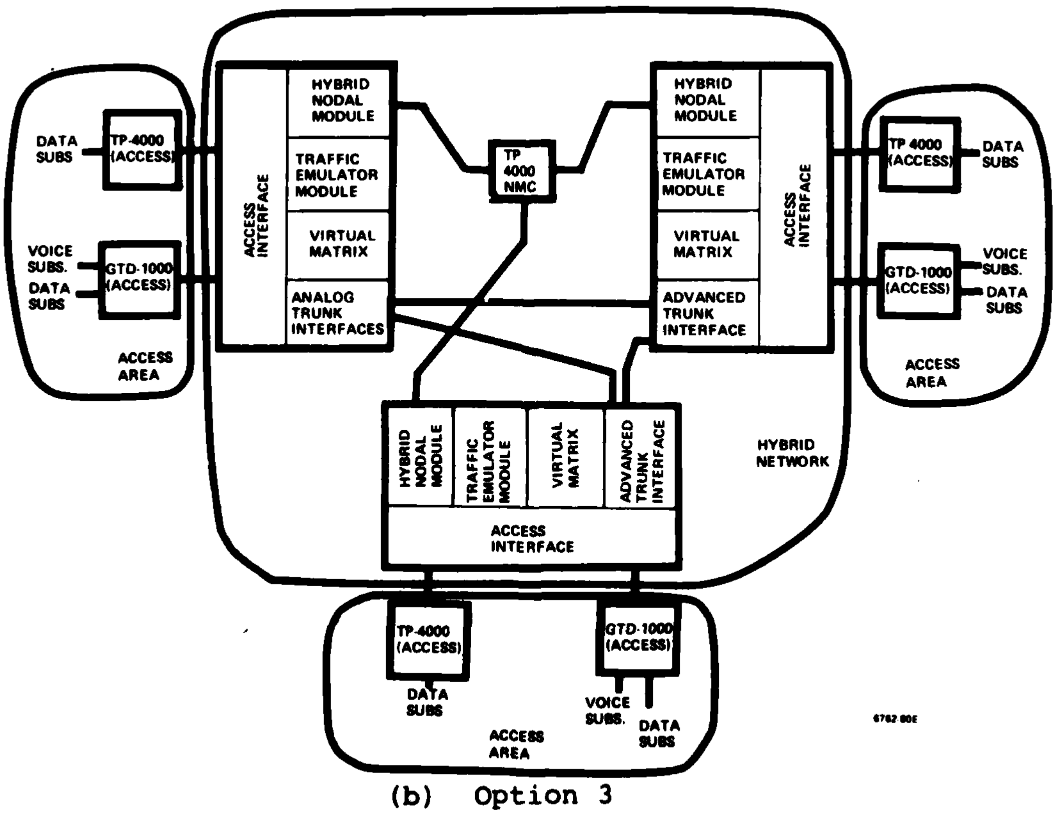
This experiment is an extension of experiment 16. The variation is that variable bandwidth circuit calls are allowed but not TASI. Due to the variable bandwidth nature of the experiment and the necessary lumping of call information into a single slot within a frame (Ref. Figure 4-13), Option I with its fixed bandwidth matrix cannot reasonably perform this experiment and will, therefore, not be coated.

Option I	Option II	Option III
Hardware Tasks:		
Not Applicable	T1 Trunk buffer Exp. Alter Access Buffers Variable Frame Matrix	T1 Trk. Buff Exp. Alter Access Buffers Variable Frame Matrix
Software Tasks:		
Not Applicable	Advance Trunk Mods Enhance NMC Statistics	(Same as II)

OPTION	SCHED. MONTHS	MATERIAL COST-$K	LABOR MAN-MOS	INSTRUCT. COUNT - K	TEST BED FIGURE
I	NA	NA	NA	NA	NA
II	8	7	57	1	7-17a
III	9	19	59	1	7-17b



(a) Option 2



(b) Option 3

Figure 7-17. Experiment 17 - Variable Frame & Boundary

7.3 Presentation of Cost/Schedule Data

The preceding analyses yielded incremental material costs, man-months of labor, and schedule in calendar months for each experiment for the three candidate architectures (Options I, II and III). Table 7-1 summarizes those results and also includes the total estimated cost, in dollars, for each of the capability steps.

Figure 7-18 presents cumulative cost/capability and schedule/capability profiles for the three options. The curves show a sequential buildup in which the capability for each step is developed in its entirety before proceeding to the next step. The curves show Option II to be somewhat more expensive ($1M to $2M) and time-consuming (10 to 20 months) than Options I and III through the first 11 experiments. The curves cross over at the hybrid baseline capability with Option II lowest by $1.6M in cost and 12 months in schedule. Essentially, however, there appear to be no great differences between the three options either in cost or schedule. For all options, the schedule shows the experiment-by-experiment sequential buildup to be an unrealistic approach. Consequently, we sought for ways to minimize the time required for implementing a full up test bed while still employing a phased build up which could yield early answers to some of the more pressing problems. These are discussed in 7.4.

7.4 ALTERNATE SEQUENCES OF EXPERIMENTS

7.4.1 Concurrent Implementation Paths

The experiment-by-experiment sequential buildup discussed above followed the relative order of experiments shown in Figure 4-1 and satisfied the experimental interdependencies shown in Figure 5-1. As noted in Paragraph 5.1, those experimental interdependencies still allowed the potential for a high degree of concurrency. The architecture of Options I and III also shows potential for concurrency inasmuch as two separate switches (one circuit, one packet) are employed at each node. The packet switching capability could be developed in parallel with the circuit switching and used to conduct the packet switching experiments (#4, #8, #15) concurrently with circuit switching experiments (#1, #2, #3, *5).

TABLE 7-1. COST/SCHEDULE SUMMARY - SEQUENTIAL BUILDUP

CAPABILITY	CKT BASE	EXP 1	EXP 2	EXP 3	PKT BASE	EXPERIMENTS 4	5	6	7	8	9	10	11	HYB BASE	EXPERIMENTS 12	13	14	15	16	17	TOTALS
OPTION I																					
Labor - MM	86	145	48	40	90	120	48	65	82	44	141	129	94	143	36	23	40	137	82	*	1593
Material - $K	145	189	104	25	528	259	29	35	46	19	91	87	140	103	25	13	88	95	52	*	2073
Total Cost - $K	743	1187	442	311	1151	1092	379	496	632	336	1080	995	802	1099	281	180	372	1049	633	*	13260
Schedule - MO	6	10	5	6	7	10	8	8	10	6	14	14	11	12	4	4	6	12	9	*	162
OPTION II																					
Labor - MM	310	100	60	32	295	101	45	36	67	40	112	120	175	-	22	23	45	105	83	57	1728
Material - $K	154	91	76	8	151	50	10	-	18	7	11	7	66	-	1	-	66	8	16	7	757
Total Cost - $K	2287	791	498	243	2165	751	352	262	506	298	807	854	595	-	160	167	387	739	615	421	12898
Schedule - MO	16	7	7	6	21	7	8	6	10	6	12	13	8	-	3	4	6	9	12	8	169
OPTION III																					
Labor - MM	86	145	48	40	90	120	48	65	82	44	141	129	94	224	43	23	40	137	93	59	1751
Material - $K	145	189	104	25	528	259	29	35	46	20	92	87	140	217	27	13	87	96	46	19	2204
Total Cost - $K	743	1187	442	311	1152	1092	380	496	632	337	1080	994	802	1756	344	180	371	1050	713	449	14511
Schedule - MO	6	10	5	6	7	10	8	8	10	6	14	14	11	15	8	4	6	12	12	9	181

*EXPERIMENT 17 NOT INCLUDED IN OPTION I

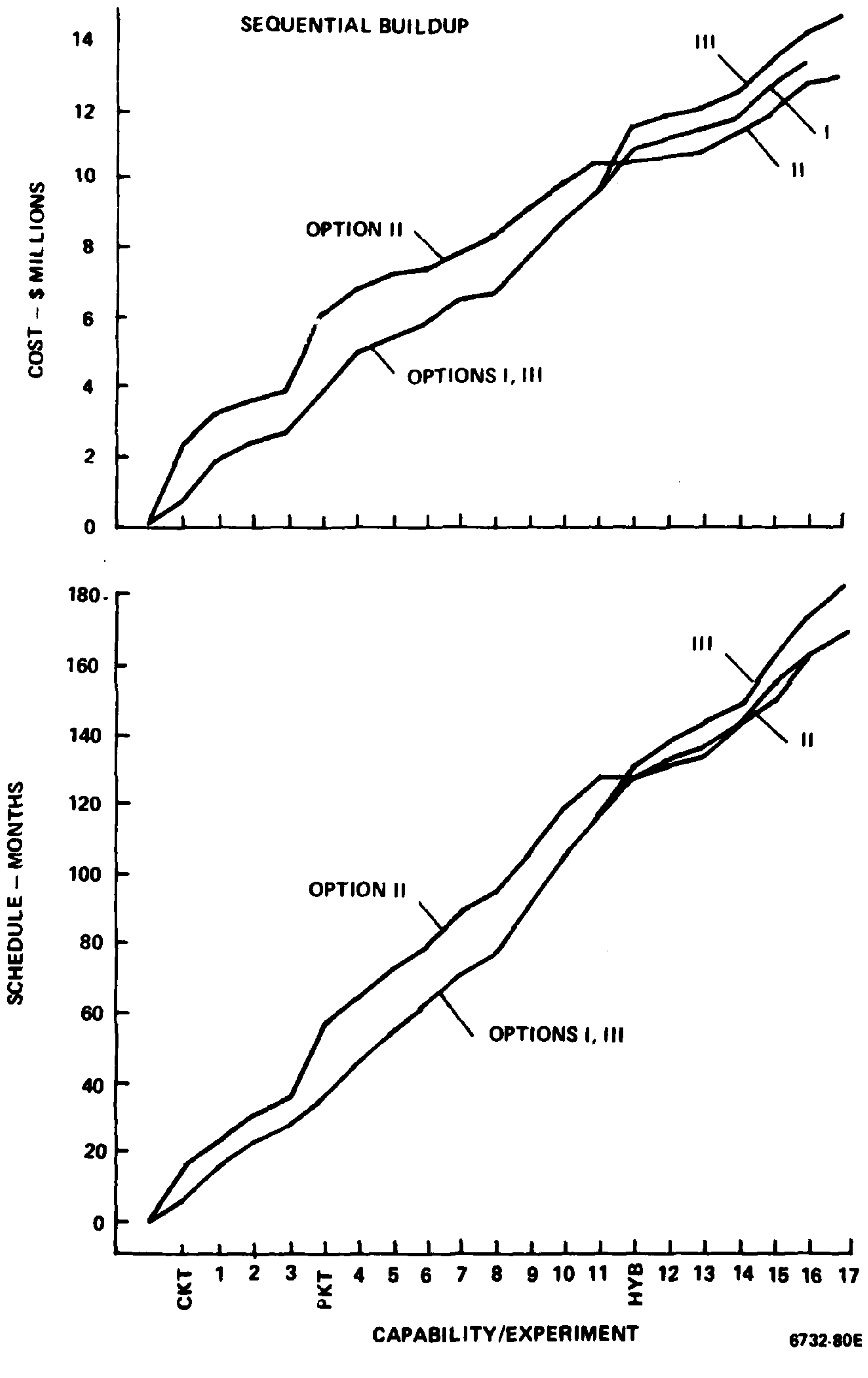


Figure 7-18. Cost/Schedule Profiles - Sequential Buildup

Table 7-2 shows the costs and schedule data for Options I and III with maximum concurrency. Cumulative cost/capability and schedule/capability profiles are shown in Figure 7-19. Note that the data for Option II is included in both the table and the figure for comparison purposes, even though a concurrent buildup isn't possible since that option doesn't employ separate circuit and packet switches. Although this approach shortens the overall schedule by almost three years, the overall program length would still be in the 10-12 year range. The costs would also decrease somewhat ($0.6M) reflecting the shorter schedule.

7.4.2 Grouping of Experiments

The test bed approaches considered thus far have involved an experiment-by-experiment buildup of capabilities. The lengthy schedules presented in Figures 7-18 and 7-19 reflect the unsuitability of such an incremental buildup. The schedule could be enhanced by judicious grouping of experiments, thus permitting parallel efforts on developing capability for more than one experiment in a given buildup step. The concurrent implementation paths described in 7.4.1 would also be implemented.

This approach was analyzed with the results summarized in Table 7-3. The cumulative cost/schedule profiles shown in Figure 7-20 indicate the specific grouping of experiments into individual buildup steps. This approach shortens the overall schedule by 4-6 years, bringing the overall program length into the 8-10 year range, depending on the option. The costs also decrease, due to the shorter schedule, down to the $12-13.5M range.

The grouped experiment approach, like the sequential and concurrent implementation path approaches, evolves to a fullup test bed retaining all of the capabilities required for the various experiments. A variation of Option III was considered in which two concurrent test beds are developed, but do not merge into one test bed having all of the capabilities. This variation is, in fact, the recommended test bed approach, and is described in Section VIII.

TABLE 7-2. COST/SCHEDULE SUMMARY - SEQUENTIAL BUILDUP & MAXIMUM CONCURRENCY

CAPABILITY	CKT BASE	EXP 1	EXP 2	EXP 3	PKT BASE	EXPERIMENTS 4	5	6	7	8	9	10	11	HYB BASE	EXPERIMENTS 12	13	14	15	16	17	TOTALS
OPTION I																					
Labor - MM	86	145	48	40	76	100	48	65	82	32	141	129	94	143	36	23	40	113	82	*	1523
Material - $K	145	189	104	25	528	259	29	35	46	19	91	87	140	103	25	13	88	95	52	*	2073
Total Cost - $K	743	1187	442	311	1036	927	379	496	632	237	1080	995	802	1099	281	180	372	851	633	*	12683
Schedule - MO	6	10	5	6	7‡	10‡	8	8	10	6‡	14	14	11	12	4	4	6	12‡	9	*	127
OPTION II						NO CONCURRENCY POSSIBLE															
Labor - MM	310	100	60	32	295	101	45	36	67	40	112	120	175	-	22	23	45	105	83	57	1728
Material - $K	154	91	76	8	151	50	20	-	18	7	11	7	66	-	1	-	66	8	16	7	757
Total Cost - $K	2287	791	498	243	2165	751	352	262	506	298	807	854	595	-	160	167	387	739	615	421	12898
Schedule - MO	16	7	7	6	21	7	8	6	10	6	12	13	8	-	3	4	6	9	12	8	169
OPTION III																					
Labor - MM	86	145	48	40	76	100	48	65	82	32	141	129	94	224	43	23	40	113	93	59	1681
Material - $K	145	189	104	25	528	259	29	35	46	19	91	87	140	217	27	13	87	96	46	19	2202
Total Cost - $K	743	1187	442	311	1037	927	380	496	632	238	1080	994	802	1756	344	180	371	852	713	449	13934
Schedule - MO	6	10	5	6	7‡	10‡	8	8	10	6‡	14	14	11	15	8	4	6	12‡	12	9	146

* Experiment 17 not included in OPTION I
‡ Done concurrently with other buildup steps

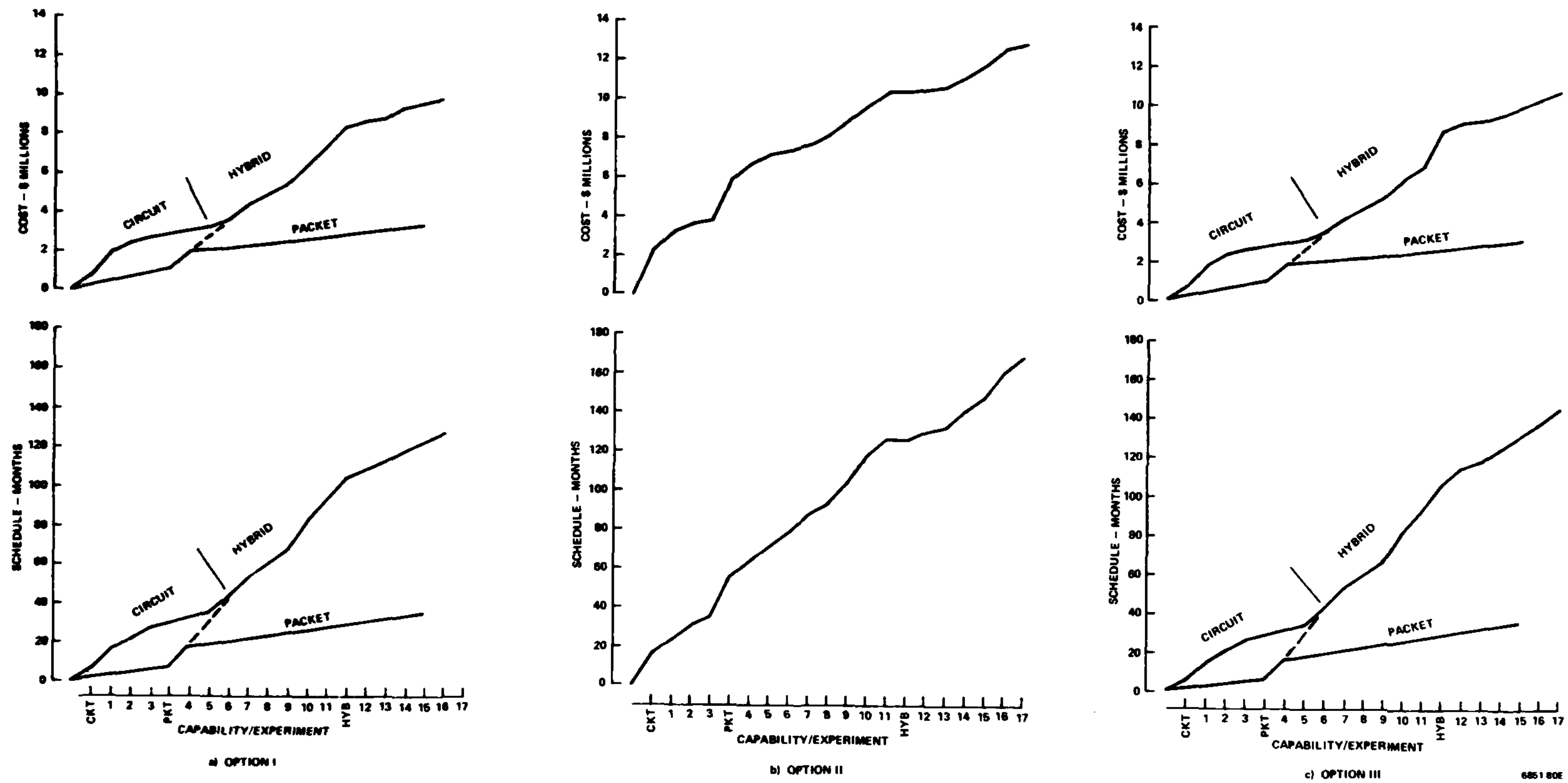


Figure 7-19. Cost/Schedule Profiles - Sequential Buildup & Maximum Concurrency

TABLE 7-3. COST/SCHEDULE SUMMARY - EXPERIMENT GROUPING & MAXIMUM CONCURRENCY

CAPABILITY	STEPS 1	2	3	4	5	6	7	8	TOTALS
OPTION I									
Labor - MM	313	101	118	374	190	82*	176	113	1467
Material - $K	464	63	66	407	143	52*	787	96	2078
Total Cost - $K	2634	775	903	3002	1462	633*	1964	852	12225
Schedule - MO	24	10	12	30	14	9*	17‡	12‡	99
OPTION II NO CONCURRENCY POSSIBLE									
Labor - MM	519	424	95	322	161	105			1626
Material - $K	349	201	25	149	24	8			632
Total Cost - $K	3940	3114	705	2395	1163	739			12056
Schedule - MO	30	30	10	24	15	9			118
OPTION III									
Labor - MM	313	101	118	374	282	136*	176	113	1613
Material - $K	464	63	66	407	257	64*	787	96	2204
Total Cost - $K	2634	775	903	3002	2214	1029*	1965	852	13374
Schedule - MO	24	10	12	30	23	13*	17‡	12‡	112

* Experiment 17 not included in OPTION I
‡ Done concurrently with other buildup steps

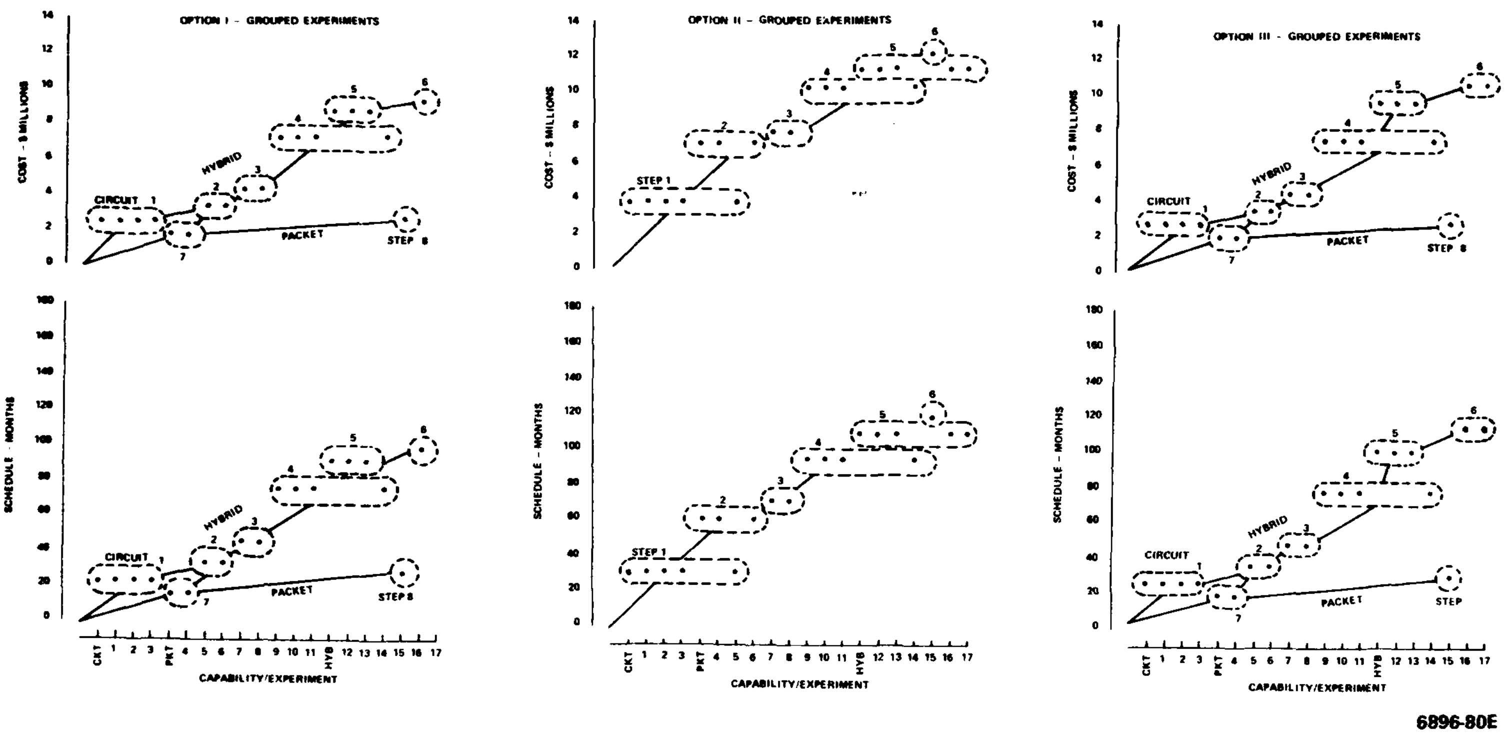


Figure 7-20. Cost/Schedule Profiles - Experiment Grouping & Maximum Concurrency

SECTION VIII

TEST BED APPROACH, EVALUATION AND RECOMMENDATION

8.1 INTRODUCTION

In Section VII, three specific candidate test bed architectures were formulated and incremental estimates of cost and schedule were generated for each of the 17 experiment groups. The result of this analysis showed only a slight difference between the three options considered and a schedule that was particularly long in terms of the rapid evolution of military equipment today. One of the ground rules was the assumption that each experiment was done sequentially, i.e., a subsequent experiment would not start until the preceding one had been completed. We realized that his step-by-step buildup of experimental capability results in an unrealistic schedule for this application. Therefore, we sought for ways in which to minimize the time required to implement the fullup test bed. By employing the methods of concurrent implementation paths (implementation of two separate experiments at the same time) and grouping of experiments, we were able to reduce the schedule considerably. In this section, we will describe the recommended test bed approach which uses the results of the optimization process to reduce the schedule by about a half.

8.2 RECOMMENDED TEST BED APPROACH

The recommended test bed approach is a variation of Option III with two separate test beds: an off-the-shelf circuit switch testbed using a GTD-1000 circuit switch, and a packet/hybrid test bed composed of a discrete buildup of distributed hardware/software processing modules. Maximum use is made of concurrent implementation and grouping of experiments in order to minimize schedule. Figure 8-1 takes the experiments identified in Figure 5-1 and shows which experiments are associated with the circuit switch test bed (dotted line) and which are associated with the packet/hybrid test bed (solid line). In this model, all experiments are performed; however, neither test bed when it has evolved to its fullup version has all the

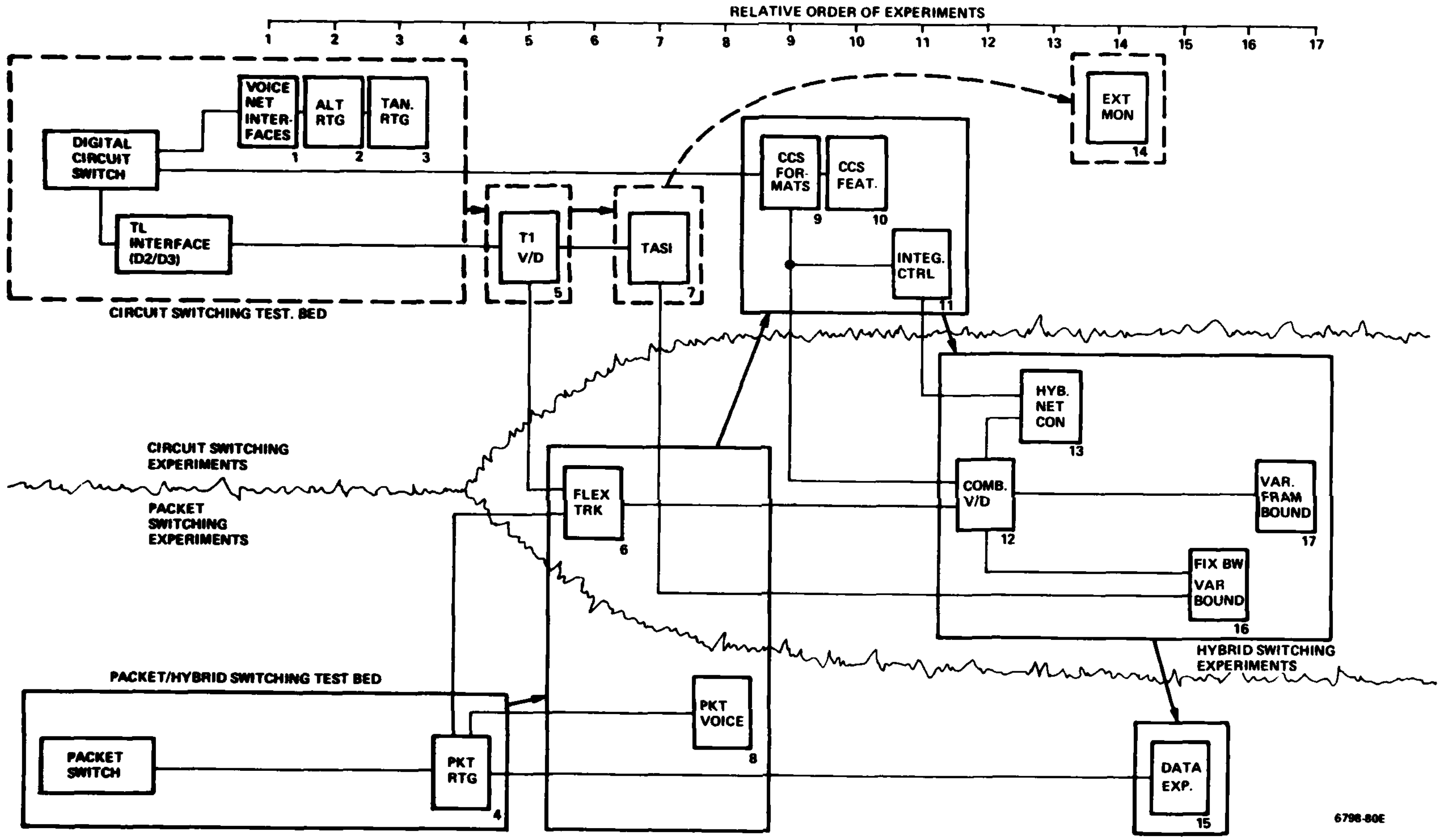


Figure 8-1. Grouping of Experiments Within the Circuit and Packet/Hybrid Switching Test Beds

capabilities of the other. For example, the packet/hybrid test bed has only enough of a circuit switching capability needed to perform the hybrid experiments. The extensive circuit switching features associated with the GTD-1000 of the circuit switching test bed are used for experiments of near-term importance, whereas the packet/hybrid test bed is used for experiments of far-term importance.

8.3 COST/SCHEDULE ANALYSIS

In costing out the recommended configuration, assumptions were made similar to those specified in Section 7.1. Costs do not include any preliminary system studies required prior to implementing an experiment, nor do they include the costs of actually performing the experiment. The test bed is assumed to be available during the hardware/software assembly and system integration and test (SIT) phases of each experimental step. Standard, commercial grade practices for hardware construction and documentation will be used. Figure 8-2 shows the cost and schedule profiles for the recommended test bed. Plotted are (a) cumulative costs vs. experiment, (b) cumulative schedule vs. experiment, and (c) cumulative cost vs. schedule for both the circuit and packet/hybrid switching test beds. The horizontal axis shows the 17 experiment groups plus the circuit, packet and hybrid switch baselines needed to perform the various experiments. Each test bed profile is broken up into steps, within which are grouped certain experiments. For example, step 1 of the circuit test bed includes implementation of a circuit switch baseline plus the performance of experiments 1, 2 and 3. Step 1 of the packet/hybrid test bed includes implementation of a packet switch baseline plus performance of experiment 4. Each of the overall steps corresponds to a specific configuration of the test bed during a phased buildup. There are four such steps for the circuit test bed and five for the packet/hybrid test bed. This will be taken up in the next section.

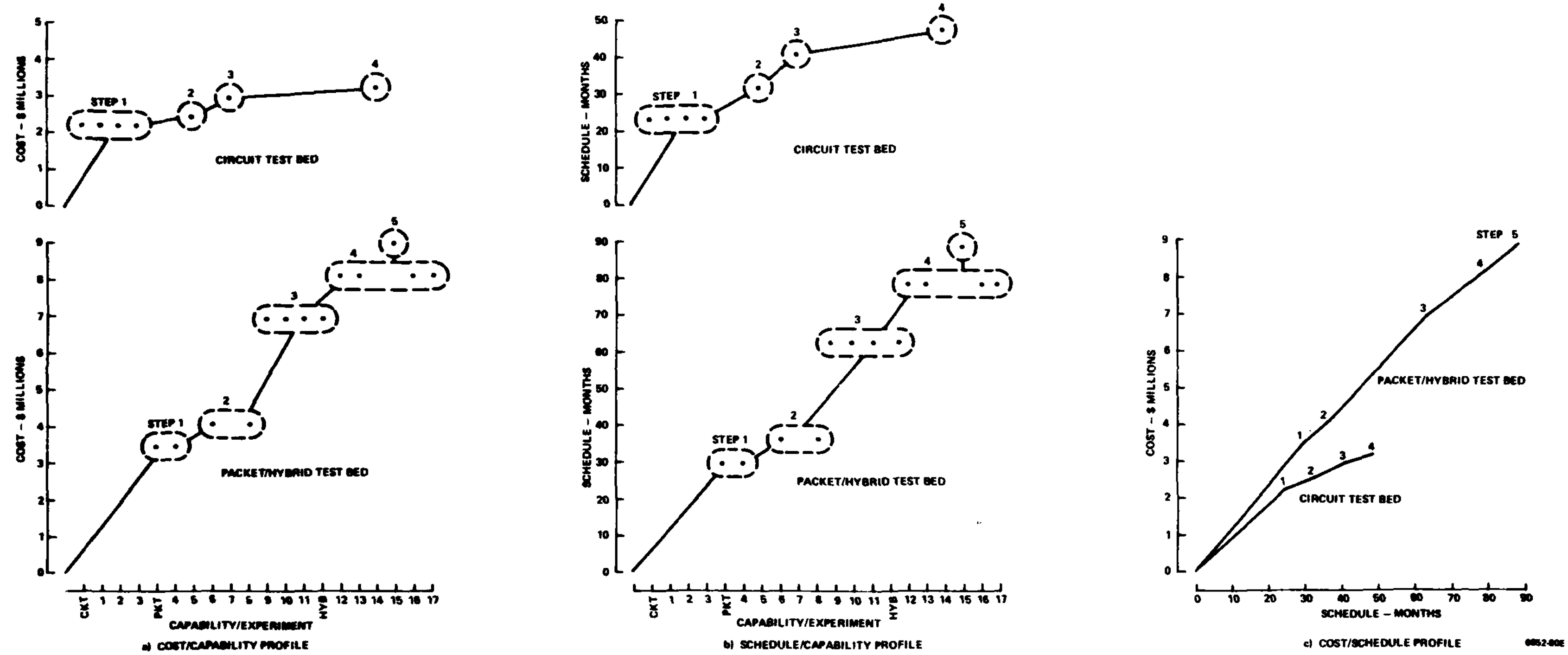


Figure 8-2. Cost/Schedule Profiles for Recommended Model

A final circuit test bed implementation would thus take four years and cost $3.2M. A final packet/hybrid test bed implementation would take 7 1/2 years and cost $8.9M. Both test beds together could be completed within the 7 1/2 year frame for a total cost of $12M. Note that with the recommended circuit test bed, the first results (completion of step 1) would not be obtained until 24 months, at which time all interoperability and routing experiments for the DSN could be performed.

Table 8-1 shows the breakdown in labor, schedule and costs for each test bed as a function of buildup step. We assume that the engineering task management annd program management functions are common to both the circuit and packet/hybrid test beds; hence, these hours have been assigned arbitrarily to the packet/hybrid test bed. Note that a considerable portion of the total cost of each test bed is associated with the first step. This is consistent with the idea that a certain amount of capability is required before the first piece of useful data can be extracted from a test bed. Figure 8-3 shows the percentage allocation of total costs for both test beds as a function of engineering discipline, management, and material.

8.4 TEST BED PHASED BUILDUP

Figures 8-4 through 8-12 show block diagrams of the phased buildup of the circuit and packet/hybrid test beds. Figure 8-4 shows the configuration for Step 1 of the circuit switching test bed. GTD-1000s, interconnected by standard T1 trunks, are used to provide a baseline experimental three-node circuit switched network which is capable of performing the interoperability and routing experiments for the DSN (experiments 1, 2 and 3). Step 2 (Figure 8-5) allows the combining of voice and data on standard T1 links (experiment 5), while step 3 (Figure 8-6) introduces TASI concepts (experiment 7) for the efficient use of transmission facilities. In step 4 (Figure 8-7), the circuit switching test bed is completed with the introduction of external monitoring and automatic test control functions (experiment 14).

TABLE 8-1. COST/SCHEDULE SUMMARY - RECOMMENDED BUILDUP

	CIRCUIT TEST BED BUILDUP STEP 1	2	3	4	TOTALS	PACKET/HYBRID TEST BED BUILDUP STEP 1	2	3	4	5	TOTALS	TOTALS FOR BOTH TEST BEDS
LABOR - MM												
Software	142	7	9	16	174	266	30	168	43	49	556	730
Electrical	45	15	40	6	106	93	10	70	36	12	221	327
Systems	70	8	10	5	93	49	22	102	47	24	244	337
Mechanical	8	2	3	1	14	8	2	6	5	2	23	37
Eng/Program Mgt	-	-	-	-	-	60	12	54	32	18	176	176
Total	265	32	62	28	387	476	76	400	163	105	1220	1607
COST - $K												
Labor	1774	219	421	185	2599	3234	531	2768	1156	731	8420	11019
Material	464	29	46	88	627	278	7	125	24	8	442	1069
Total	2238	248	467	273	3226	3512	533	2893	1180	739	8862	12088
SCHED. - MONTHS												
Per Step	24	8	9	7	48	30	6	27	16	9	88	88

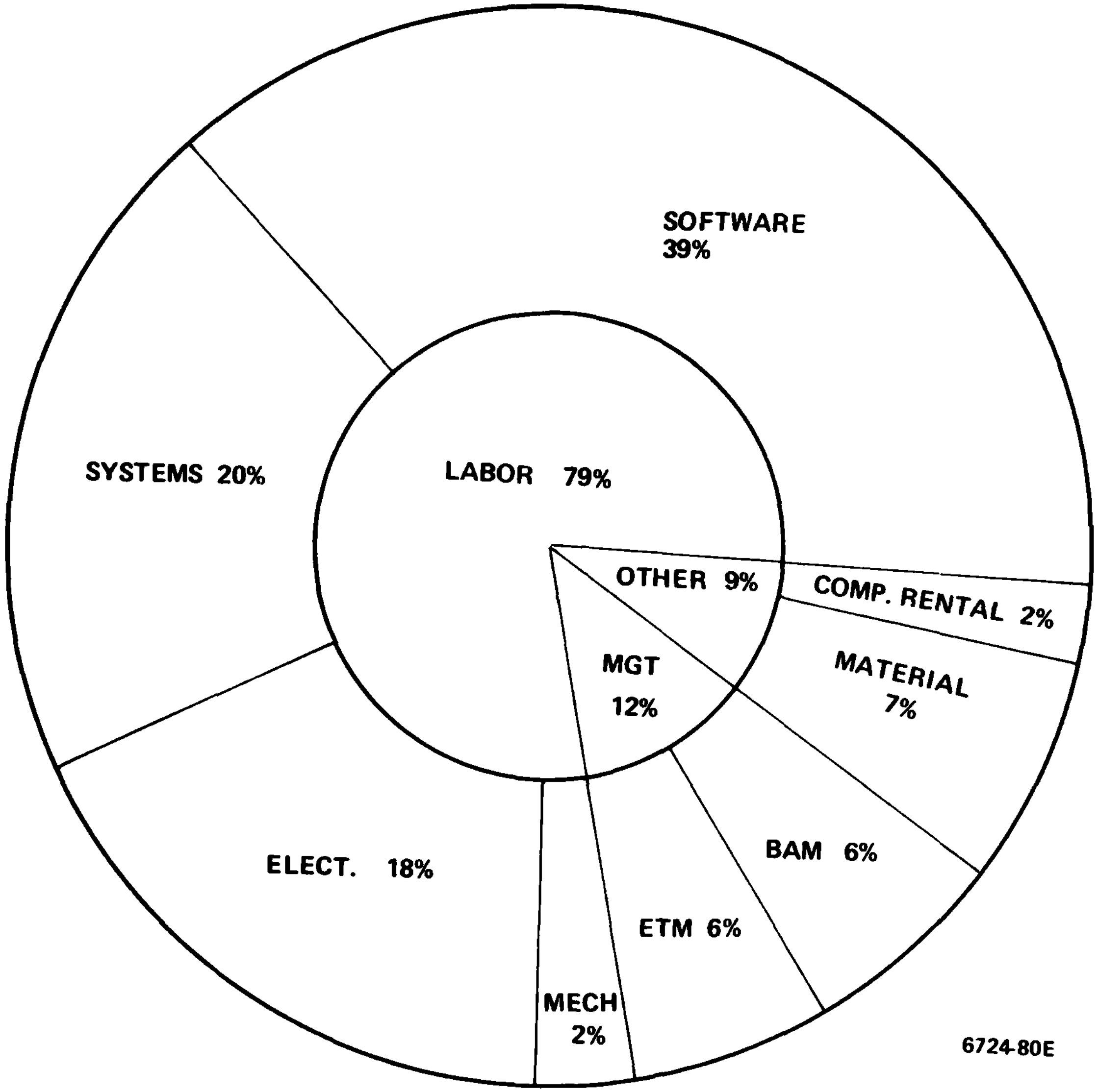


Figure 8-3. Allocation of Costs - Recommended Approach

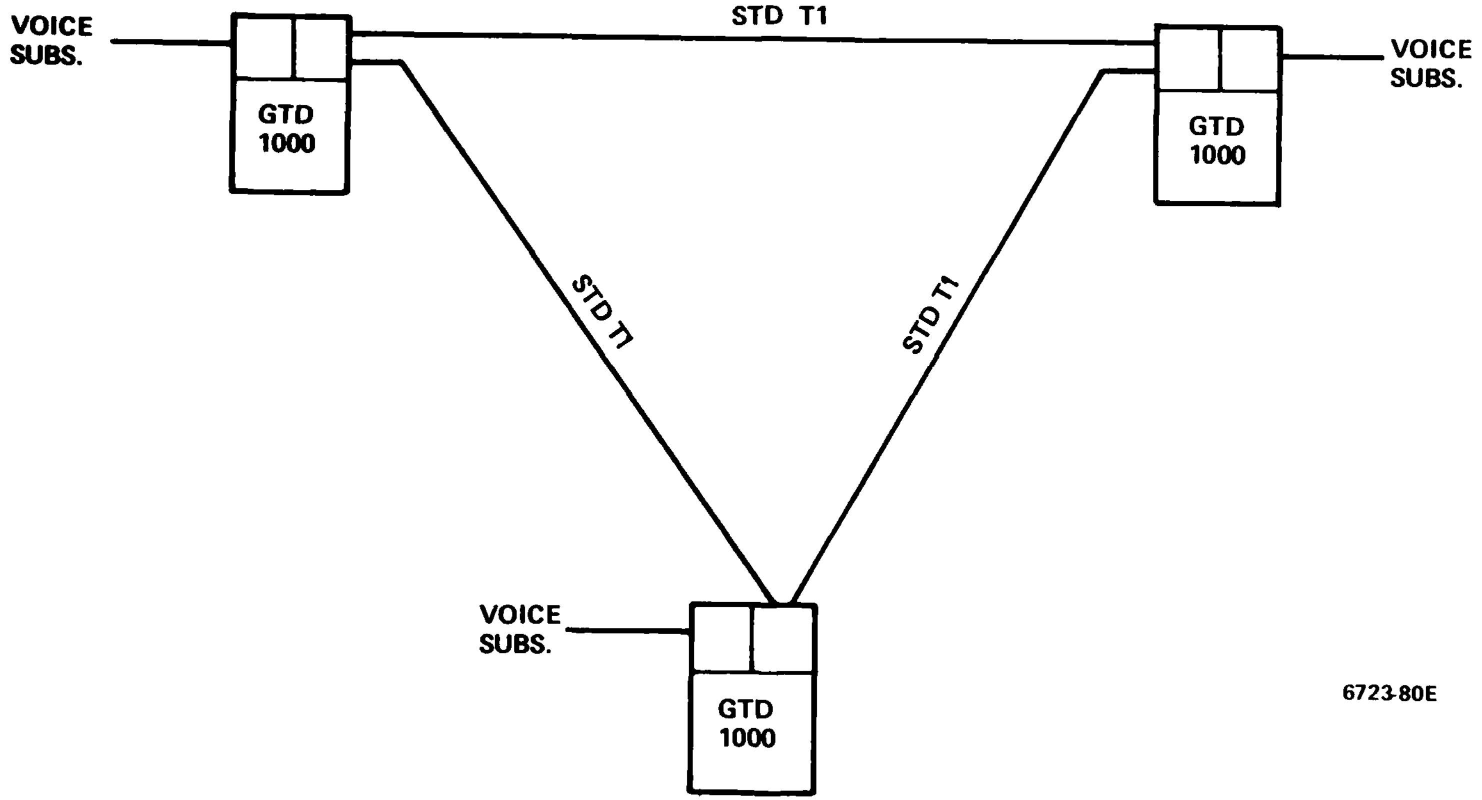


Figure 8-4. Recommended Buildup for Circuit Test Bed - Step 1

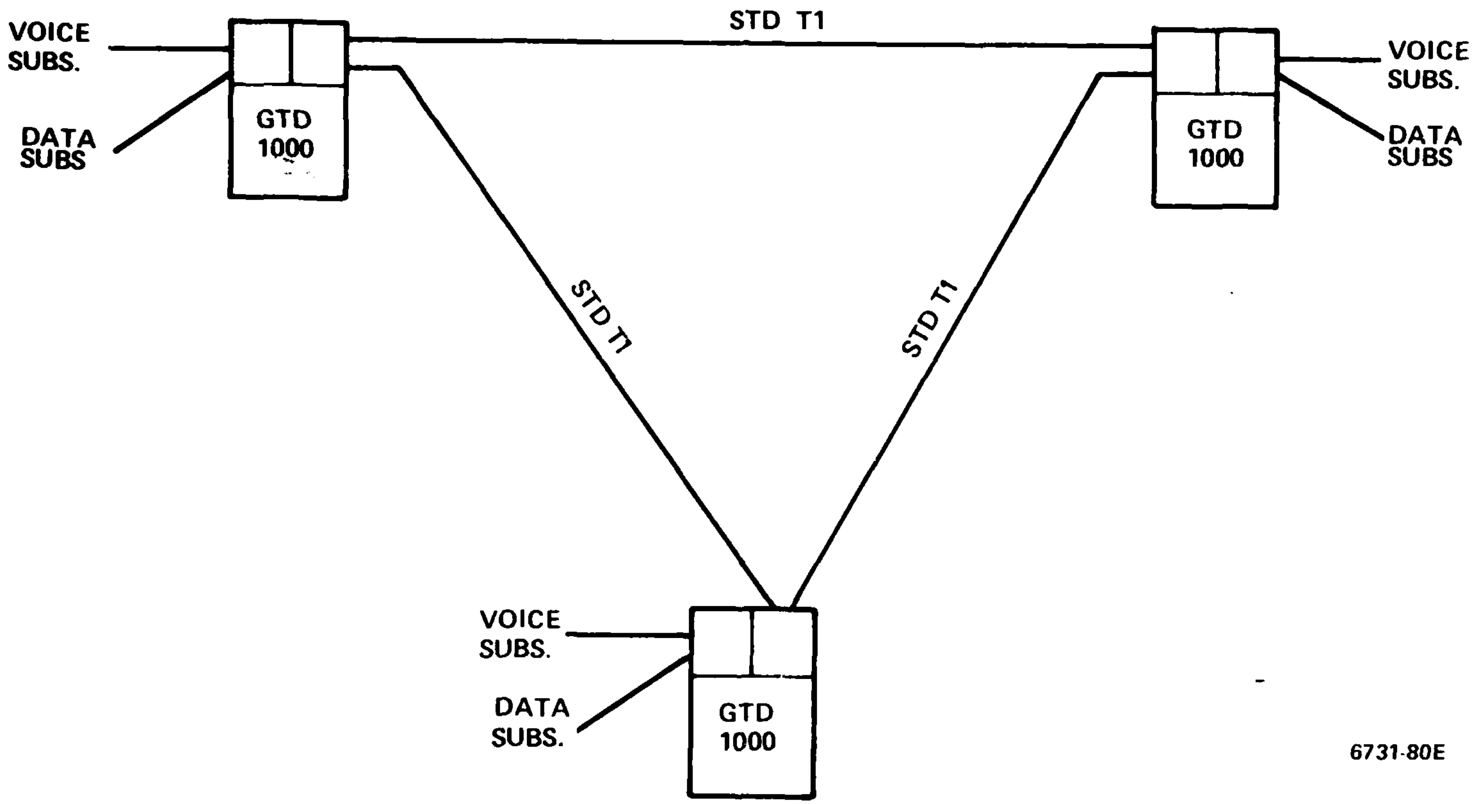


Figure 8-5. Recommended Buildup for Circuit Test Bed - Step 2

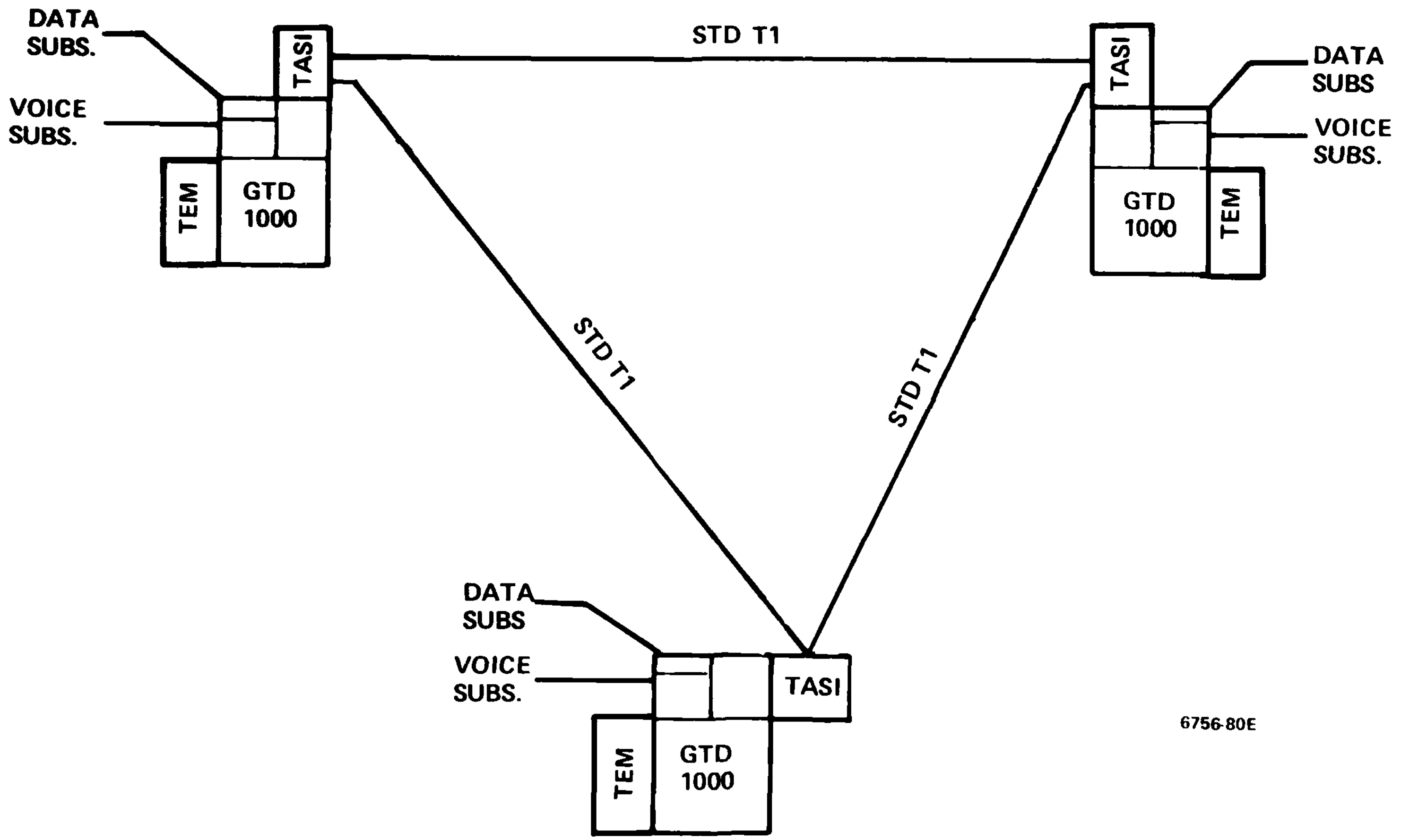


Figure 8-6. Recommended Buildup for Circuit Test Bed - Step 3

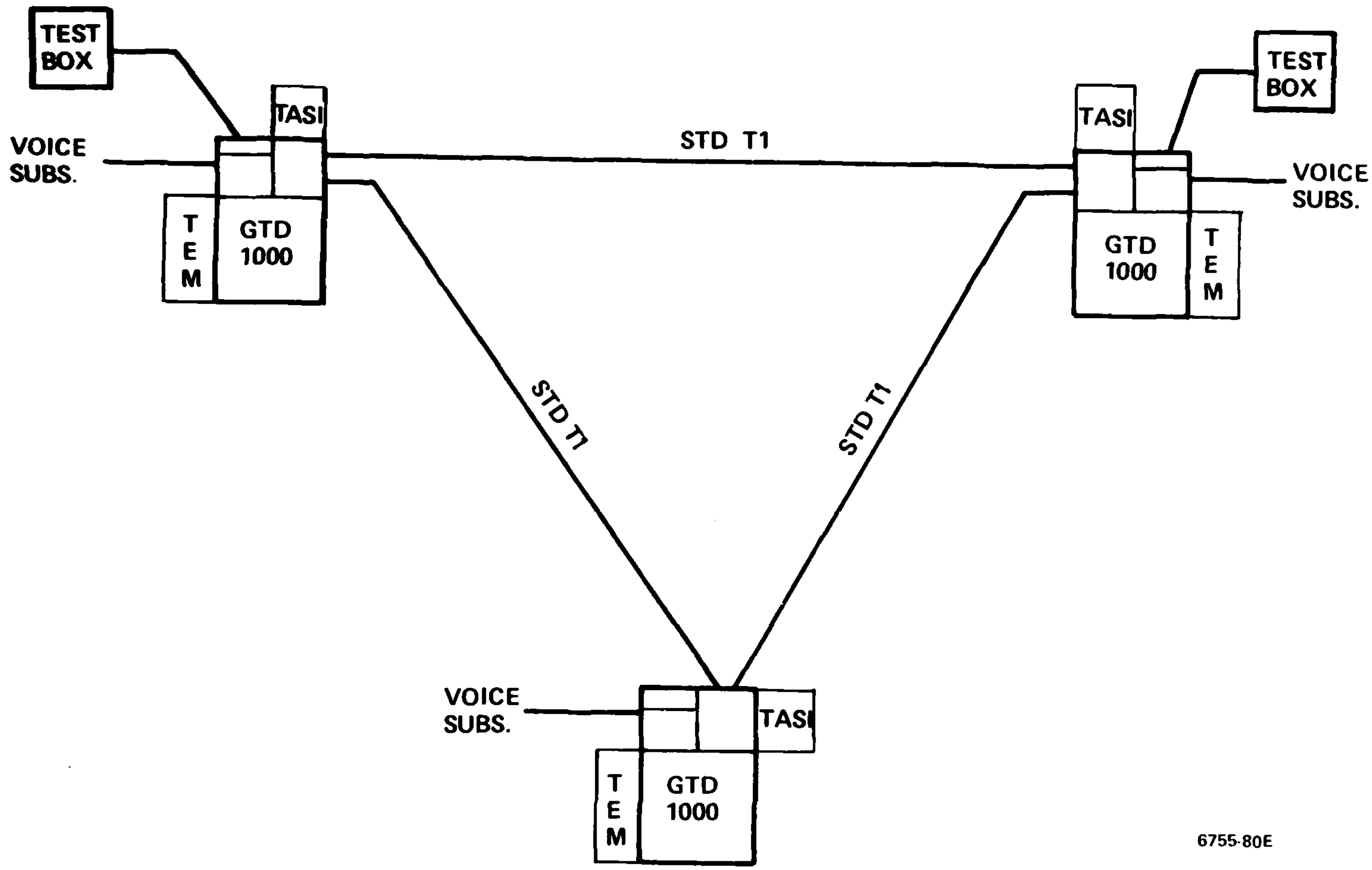


Figure 8-7. Recommended Buildup for Circuit Test Bed - Step 4

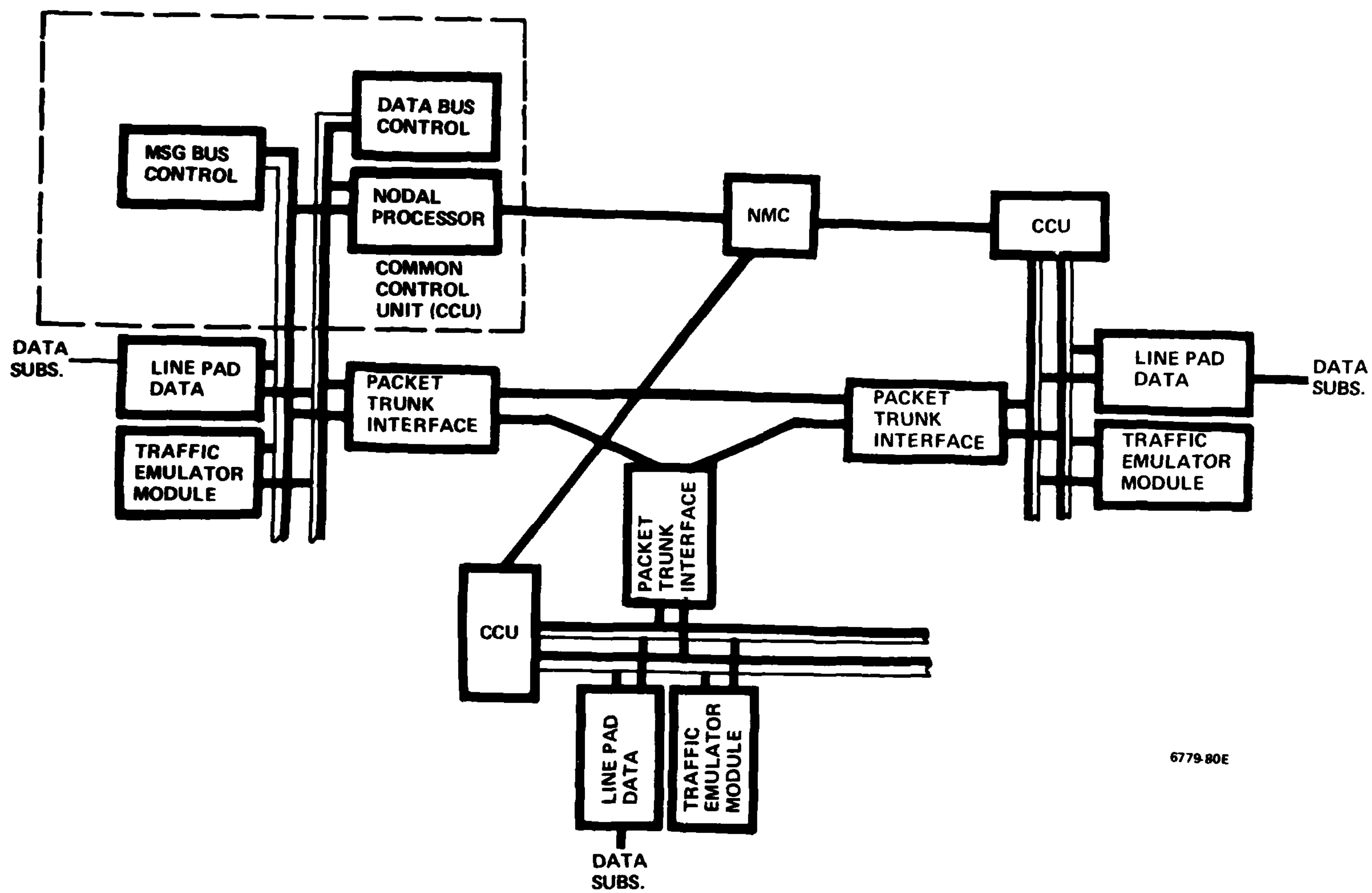


Figure 8-8. Recommended Buildup for Packet/Hybrid Test Bed - Step 1

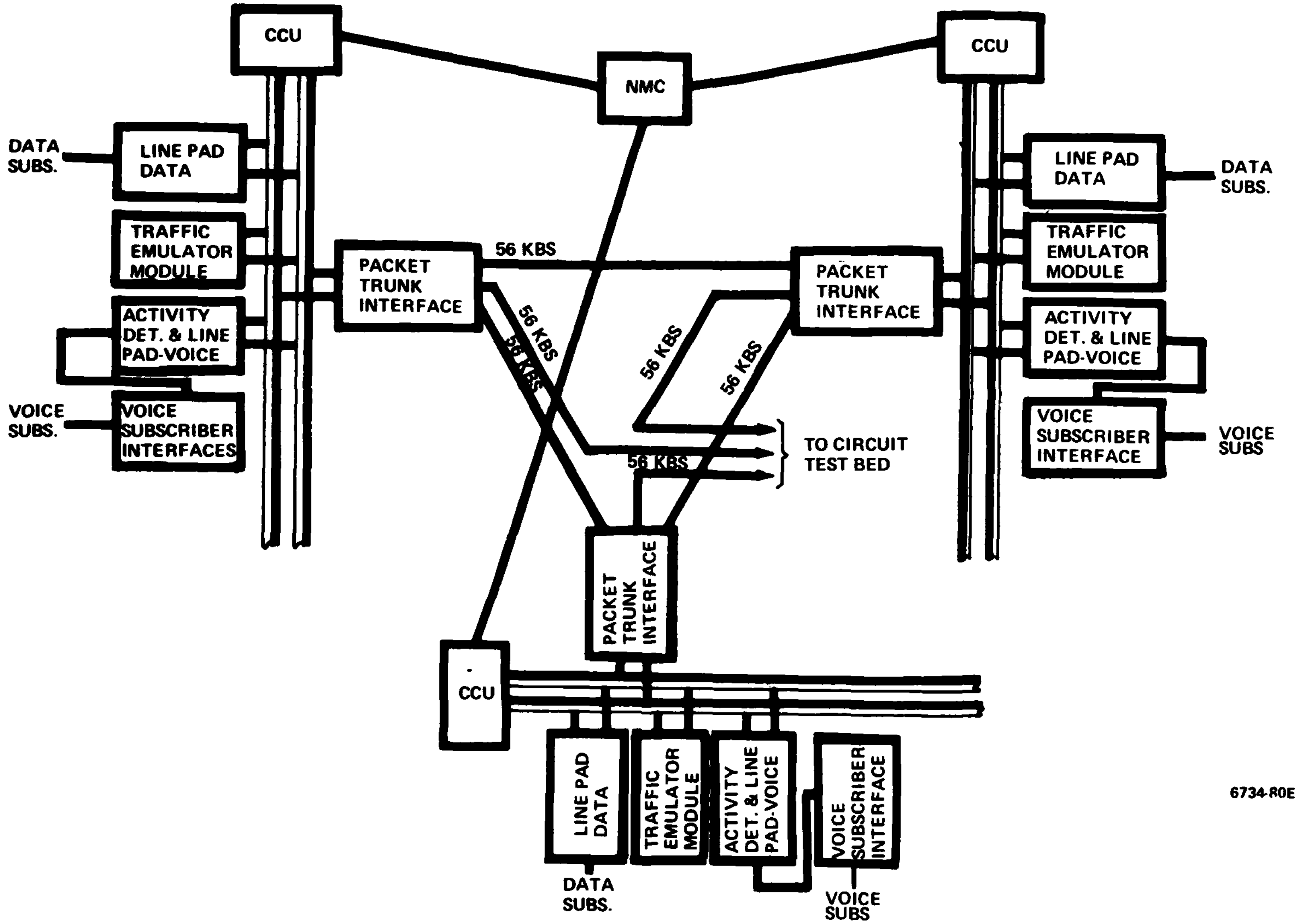


Figure 8-9. Recommended Buildup for Packet/Hybrid Test Bed - Step 2

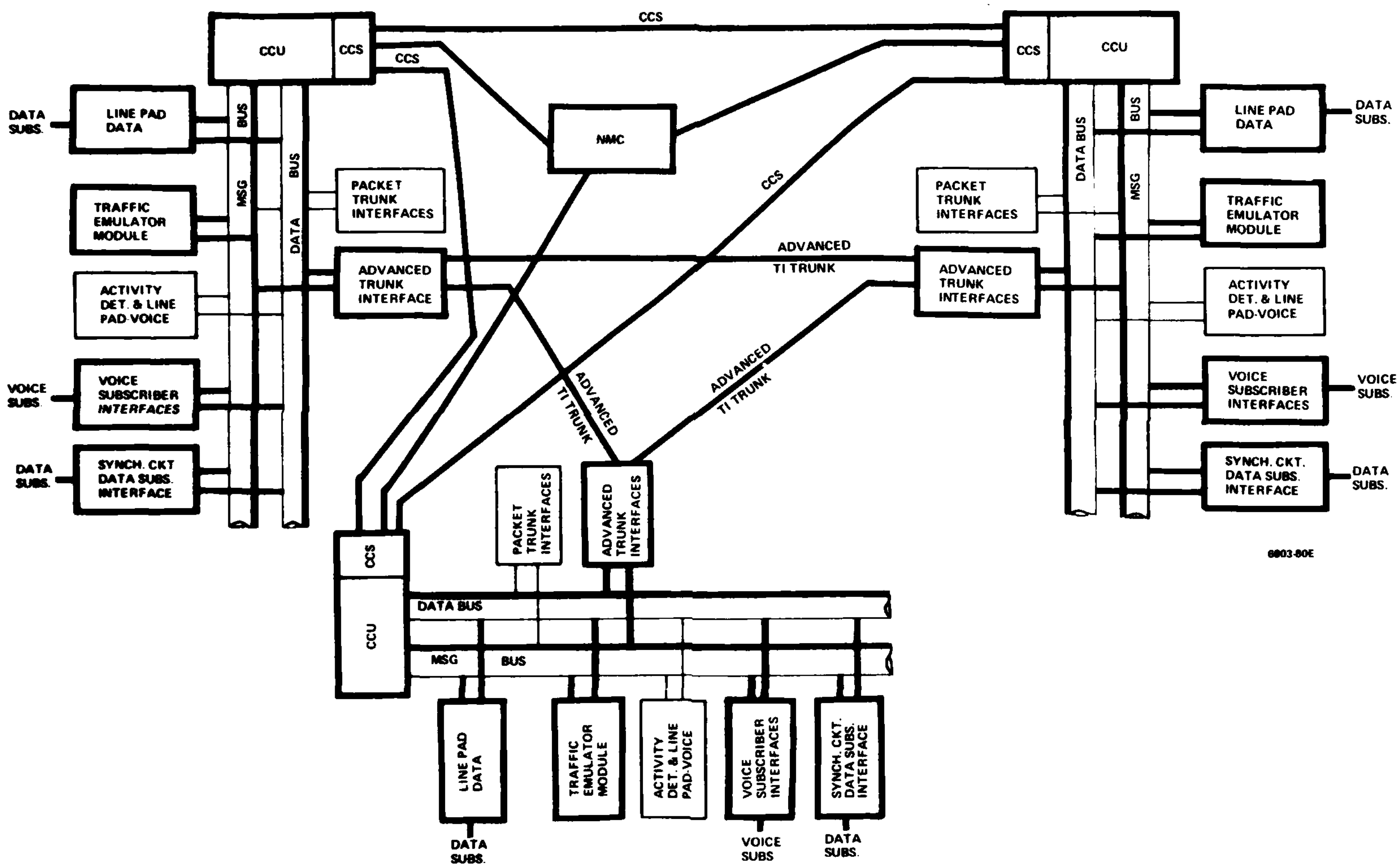


Figure 8-10. Recommended Buildup for Packet/Hybrid Test Bed - Step 3

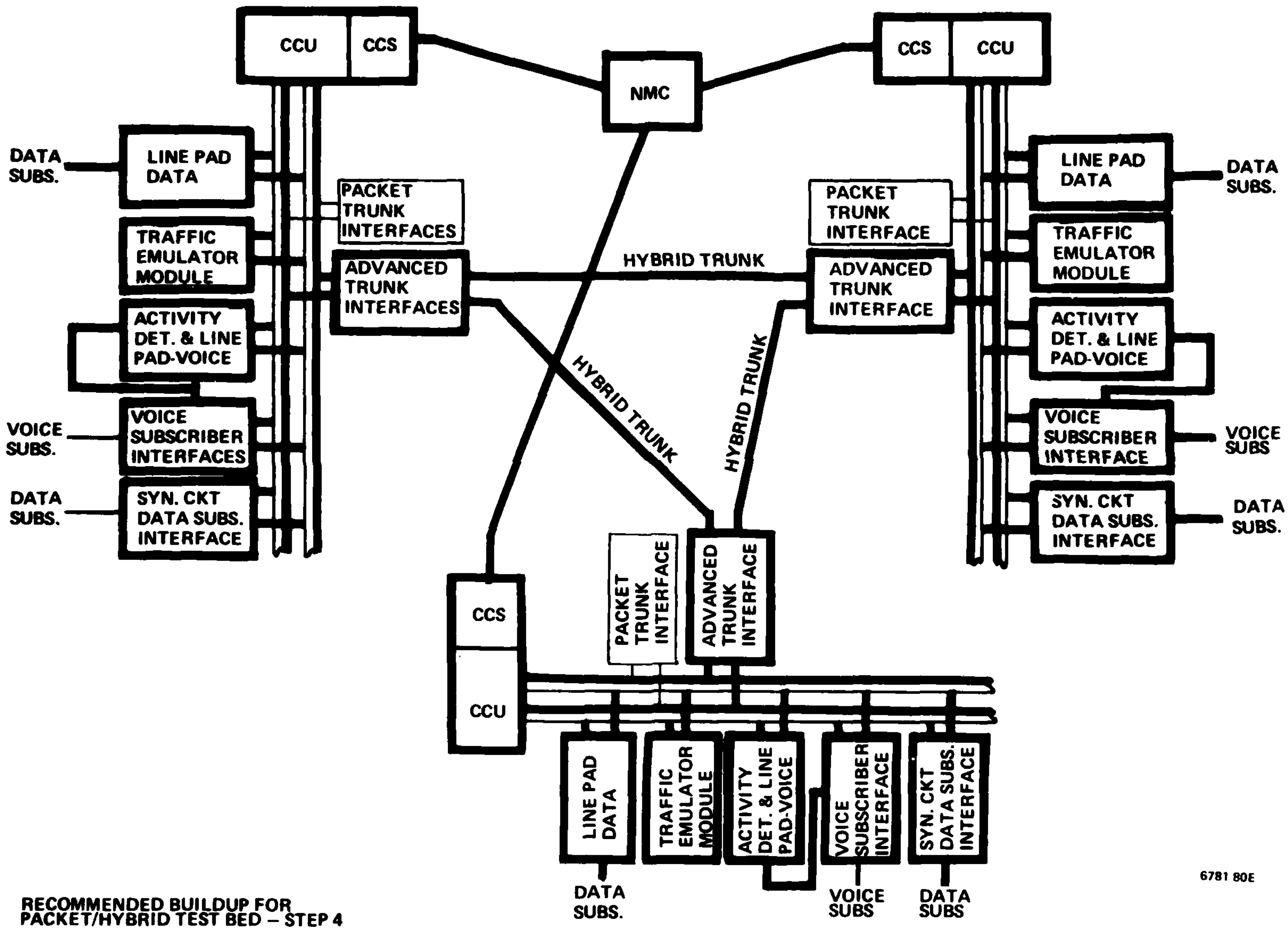


Figure 8-11. Recommended Buildup for Packet/Hybrid Test Bed - Step 4

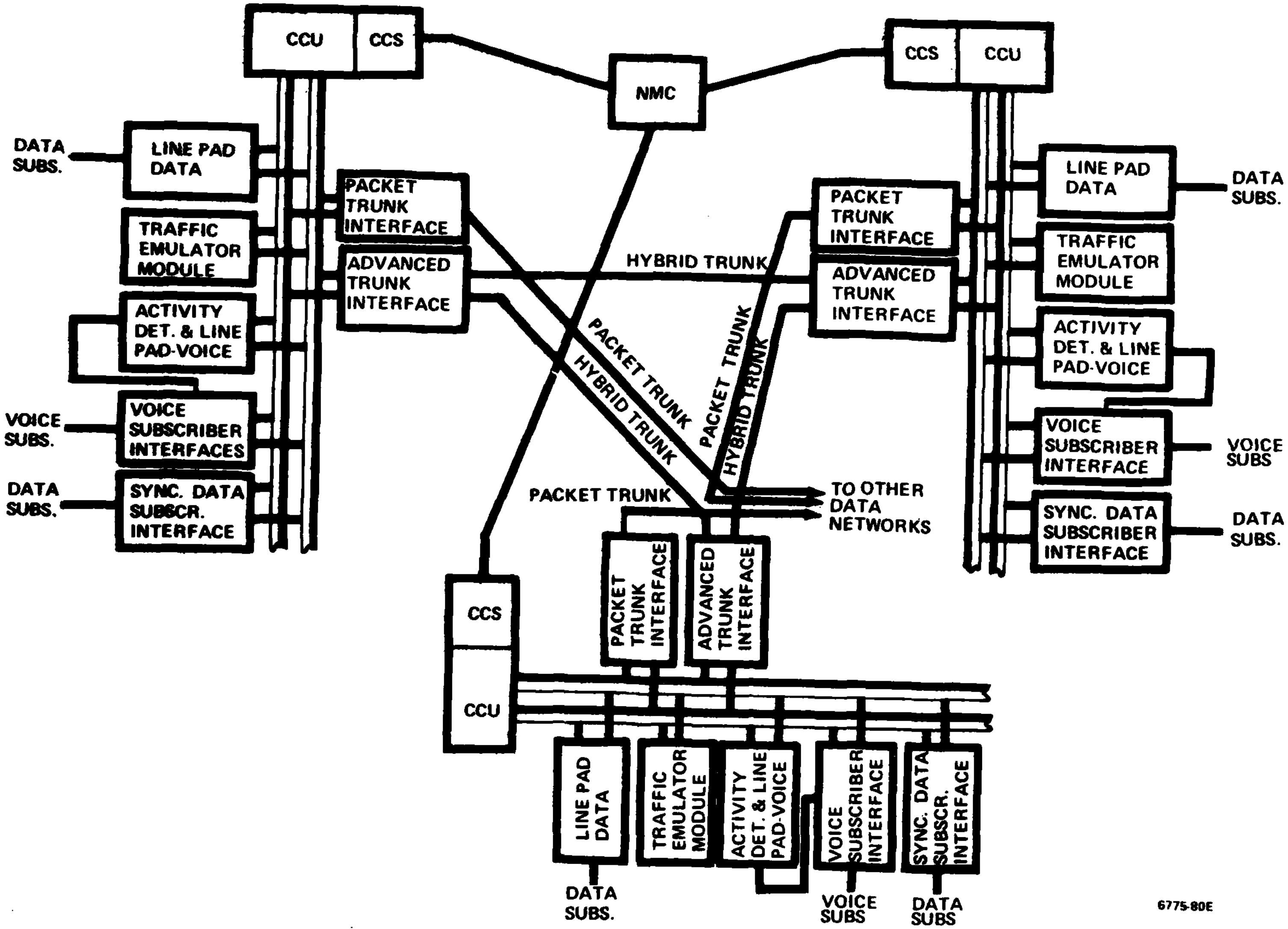


Figure 8-12. Recommended Buildup for Packet/Hybrid Test Bed - Step 5

Figure 8-8 shows a baseline experimental three-node packet switched network, composed of a distributed arrangement of packet switches interconnected by 56 KBPS trunks, and a network measurement and monitoring center (NMC). This configuration is capable of dynamic packet switch routing experiments (experiment 4) to take advantage of trunk availability. Step 2 (Figure 8-9) adds a flexible trunk capability for packet switches to offload traffic during congestion on a dynamic basis (experiment 6) and packetized voice (experiment 8). Step 3 (Figure 8-10) introduces common channel signaling (experiments 9 and 10), integrated voice/data network control, particularly during periods of contention (experiment 11) and adds a baseline hybrid (circuit/packet) switching capability. In step 4 (Figure 8-11), more sophisticated combined voice and data experiments are performed (experiment 12) and the use of hybrid switching techniques for integrating voice and data (experiments 13, 16 and 17) are investigated. Step 5 (Figure 8-12) completes the packet/hybrid test bed by introducing some advanced data experiments for the interfacing of different packet switching networks (experiment 15).

8.5 SUMMARY AND CONCLUSIONS

In this report, we have identified issues relative to the newly defined Defense Switched Network (DSN) and devised experiments to address these issues. The experiments were put in sequential order based on their immediacy and relative importance to further development of the DSN. From the definition of the experiments came an identification of the requirements for a test bed which could perform those experiments.

Subsequently, several generic architectures for a test bed which would meet these requirements were designed and specific versions of these architectures were used to perform a cost/schedule/experimental capability analysis. Table 8-2 presents the result of this analysis. It shows the results for the three approaches (options) using a sequential ordering of experiments and illustrates the iterations used to derive the recommended model for a test bed. This is the combination of circuit and packet/hybrid test beds employing maximum concurrency and grouping of experiments.

TABLE 8-2. COST/SCHEDULE SUMMARY

BUILDUP METHOD	OPTION	COST	SCHEDULE
SEQUENTIAL BY EXPERIMENT	I II III	$13.3 M* $12.9 M $14.5 M	13 Y, 6 M* 14 Y, 1 M 15 Y, 1 M
SEQUENTIAL WITH MAXIMUM CONCURRENCY	I II III	$12.7 M* $12.9 M $13.9 M	10 Y, 7 M* 14 Y, 1 M 12 Y, 2 M
GROUPED EXPERIMENTS WITH MAXIMUM CONCURRENCY	I II III	$12.2 M* $12.1 M $13.4 M	8 Y, 3 M* 9 Y, 10 M 9 Y, 4 M
RECOMMENDED	CKT TEST BED PKT/HYB TEST BED	$ 3.2 M $ 8.9 M	4Y 7 Y, 4 M

*EXPERIMENT 17 NOT INCLUDED IN OPTION I

It should be noted that these results are for test beds with full experimental capability; they perform all of the experiments described in Section 4. These include 17 experiment groups and some 63 total sub-experiments (tests). It is possible to choose a subset of these experiments and from this derive a smaller version of the test bed, with a corresponding savings in cost and schedule. However, this presumes knowledge by the designer that certain experiments are crucial (e.g., to development of the DSN) while others can be ignored or put off until a later time. Therefore, depending on the issues of critical importance, and hence, the experiments devised to address those issues, the results presented here will change. Under these conditions, cost and schedule may decrease significantly.

APPENDIX A

GTD-1000 CIRCUIT SWITCH

GTD-1000 DIGITAL PABX
FULL-LINE VERSION
SYSTEM VERSION RELEASE (SVR) 2.2.1.0
DESCRIPTION AND OPERATION

CONTENTS PAGE

1. GENERAL

1.01 This section provides a description of the GTD-1000 Digital PABX and the system's sequence of operation during the processing of a station-to-station call.

1.02 The GTD-1000 Digital PABX is a stored-program system featuring Pulse Code Modulation (PCM) switching techniques. All major call-processing functions are controlled by a software program. The system may optionally access T1-type lines for interoffice trunks.

1.03 Refer to the appropriate section of the 278 division of GTE Practices for a detailed description of the system's features and services. Section 810-278-076 provides a detailed listing of the equipment specifications, floor plans, and engineering applications.

1.04 This section is reissued to include information on the Centralized Attendant Service (CAS) option, the expansion of file D of the Common Equipment Cabinet (CEC), and to include information on the Health Care/Motel (H/M) option of the system. Marginal arrows are used to identify the new material. Remove the previous issue of this section from the binder or microfiche file and replace it with this issue.

2. DESCRIPTION

2.01 The system is housed in cabinets consisting of one CEC and as many as four Peripheral Equipment Cabinets (PEC's). An optional second CEC may be added to the basic system to provide redundant common control. Each cabinet is 72-5/16 inches high, 28-1/2 inches wide, and 25 inches deep. The cabinets can be bolted together or can stand alone. Universal doors are provided for left- or right-hand mounting. The PEC's may be no more than 50 cable feet apart.

2.02 System cabling is routed through the cabinet tops to the CDF at sites using typical central office methods. At sites equipped with computer-room-type (raised) floors, cabling can be routed through access holes in the bottom of the cabinets. Connectorized cables are provided for inter-cabinet cabling and cabling to the CDF. Plug-ended quick-connect (PABX standard) terminal blocks are available for the CDF.

Centralized Attendant Service Option

2.03 Centralized Attendant Service (CAS) is provided as an option with the system. With CAS, the various branch PABX's and the main location are interconnected by Release Link Trunks (RLT's). The RLT's route attendant-seeking traffic from the branch PABX to the CAS main location on a First In First Out (FIFO) basis. Attendant-seeking calls are automatically and uniformly distributed to the centralized attendants by Automatic Call Distribution (ACD) located at the CAS main installation. The ACD and the CAS main location are not part of the branch PABX. The RLT's release for reuse when a centralized attendant has extended the call to a trunk or station at the branch PABX. The centralized attendants use supervisory signals, address signals, and information tones between the main location and the branch PABX's to determine the type of call assistance required and to control PABX operation at the branch locations. Figure 1 shows the CAS system configuration. The exact features provided by the CAS attendant console are site-dependent.

2.04 The CAS attendants handle Listed Directory Number (LDN) calls, dial-0 calls from stations and tie trunks, transfer calls, and intercept calls. Conference, WATS, and other special-feature calls are handled locally at the branch PABX or by tie trunks. The pooled group of attendants that handles calls from the branch PABX's require two identifying signals to process calls. The signals are as follows:

(a) A console lamp or display to identify the originating branch PABX.

(b) Information tones originating at the branch PABX to identify the type of condition of a call.

2.05 To control the branch station equipment, the CAS attendant uses the following:

(a) A FLASH pushbutton to send hookswitch flash signals over the RLT's. This pushbutton sends a timed hookswitch flash (on-hook) signal by way of the facility to the RLT at the PABX.

(b) A key pad to send DTMF signals over the RLT's. The attendant uses the key pad to access a particular station or trunk at the branch PABX.

(c) A RELEASE pushbutton to release from the RLT's. The attendant depresses this pushbutton to disconnect from a call.

2.06 The branch PABX's send supervisory and information signals to the CAS equipment at the main location by means of an RLT. The main location sends supervisory and address signals to the branch PABX's by means of an RLT that can be seized only by the equipment at the branch PABX.

Health Care/Motel Service Option

2.07 The H/M option provides electronic telephone service for health care facilities and motels. The system is equipped with a Key Entry Display Unit (KEDU) that controls various H/M features that cannot be served by standard attendant or station facilities (Figure 2). The KEDU provides the following features:

(a) Wake-up (room call).
(b) Message metering.
(c) Message waiting.
(d) Unoccupied room restriction.
(e) Do not disturb.
(f) Room status display.
(g) Display of real time.

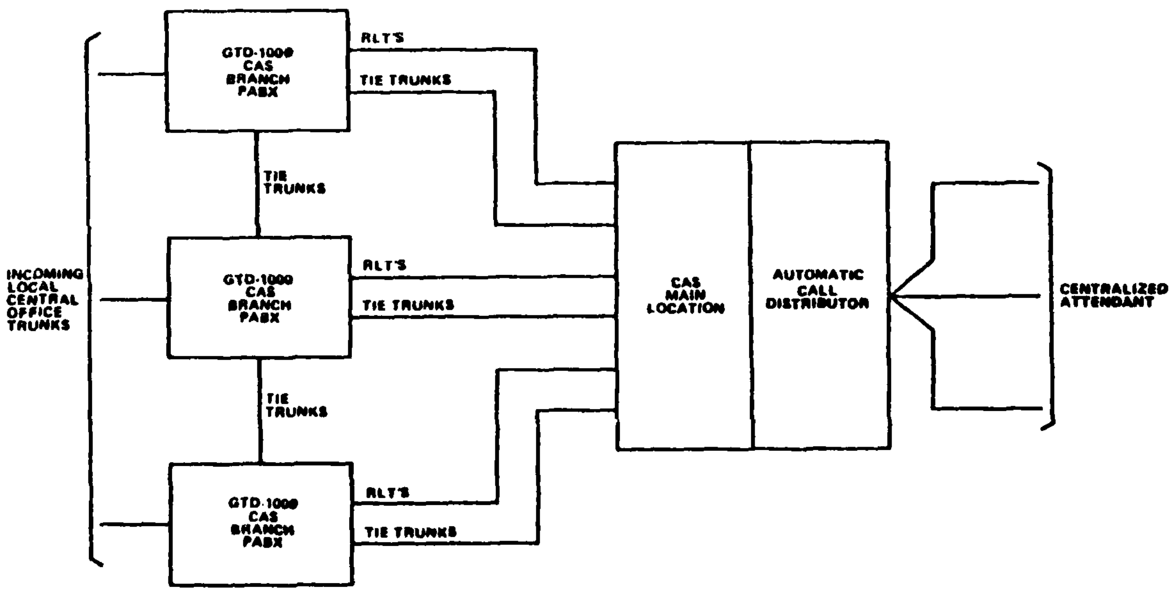


Figure 1. CAS System Configuration.

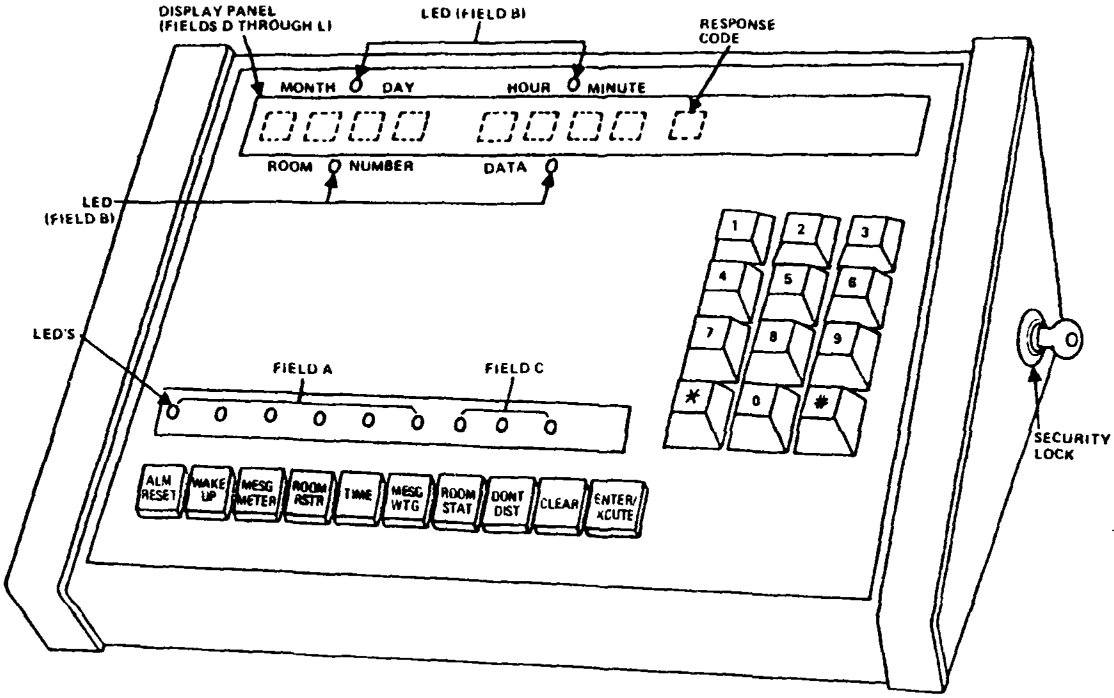


Figure 2. Key Entry Display Unit (KEDU) for H/M Option.

2.08 The KEDU is powered from a dc source (—48 Vdc output from the PEC) via a ac/dc converter within the KEDU housing. If power from the cabinet is insufficient or the KEDU is in a remote location, a local source of —48 Vdc (ac/dc converter) may be used.

2.09 The system may have up to four print-only, hard-copy printers. The printer operates at 300 or 1,200 baud and is suitable for table-top operation. The output data is data base programmed and may be any combination or all of the following:

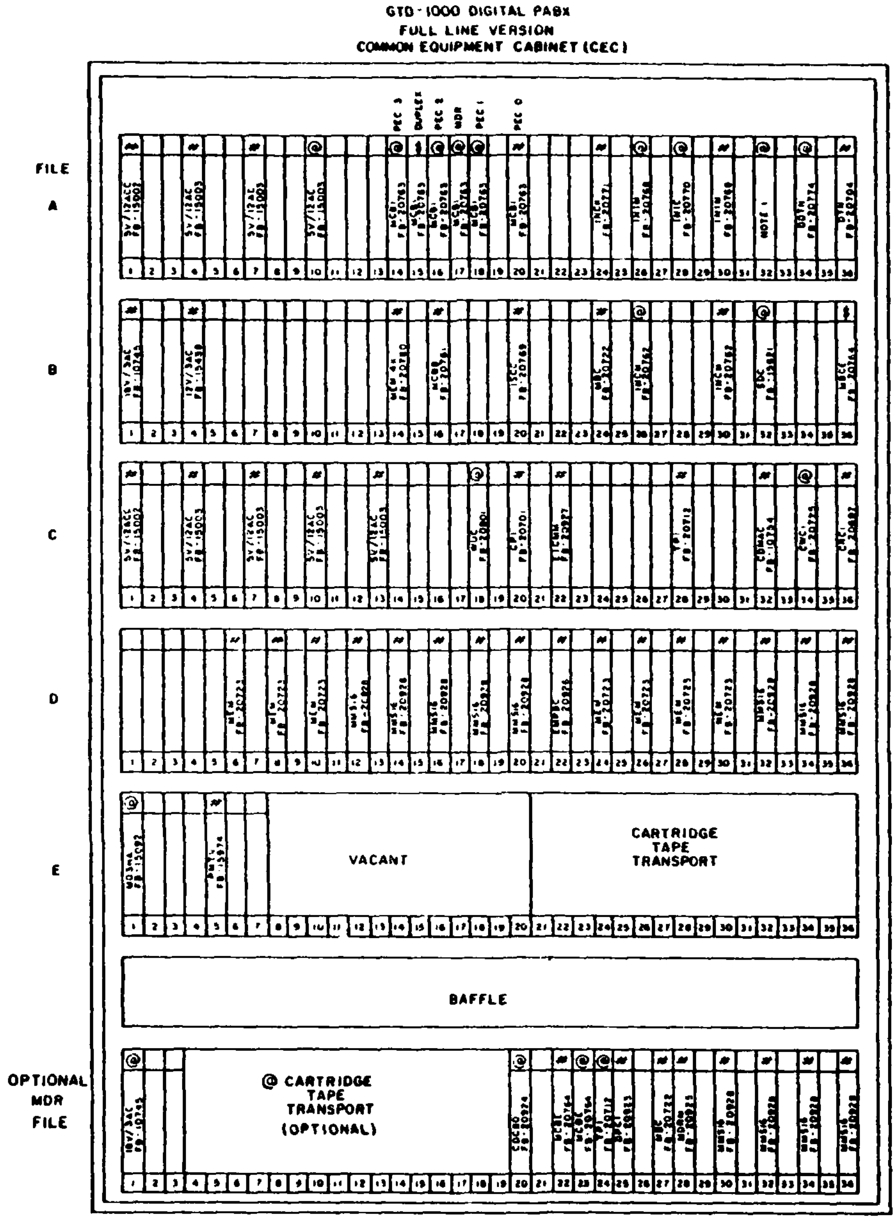
(a) Wake-up transaction data.
(b) Message waiting transaction data.
(c) Do not disturb transaction data.
(d) Room status.
(e) Room restriction.
(f) Message meters.
(g) Audit data.

System

2.10 The system uses plug-in printed wiring cards mounted in an extended Standard Hardware Electronic Systems (SHES) card file with 36 card slots per file. A CEC (Figure 3) contains 4 files of 36 card slots each and 1 file of 7 card slots. A PEC (Figure 4) contains 4 files of 36 card slots each with 10 slots available for lines, trunks, and feature cards. A fifth file contains the low voltage power cards. The CAS option is implemented by inserting new cards into the card files, as shown in Figures 3 and 4. The H/M option is shown in Figures 5 and 6.

2.11 Optional features are incorporated and system expansion is accomplished by adding cards to the system. Expansion of the system from 256 lines to 1,000 lines is accomplished by adding equipped PEC's (maximum of 4). This modular approach allows a system designed to a specific level to be upgraded as necessary without disrupting the basic system concept.

2.12 System cabinets are designed with cutouts and louvers for cooling. The cabinet openings, the open card file design, and a high-efficiency power supply result in a convection-cooled installation that does not require fans. The maximum weight of the system is approximately 3,600 pounds. The floor loading capacity is less than 125 pounds per square foot.



\# - DENOTES ALL BASIC PACKAGE PWC'S ALWAYS PROVIDED WITH THE FULL LINE COMMON EQUIPMENT CABINET (EC-21932-A).

@ - DENOTES PWC'S AND EQUIPMENT REQUIRED FOR OPTIONAL FEATURES AND/OR ADDITION OF PEC CABINETS.

$ - DENOTES PWC'S LOCATED IN THESE SLOTS COME EQUIPPED WITH ALL DUPLEX BASIC PACKAGES. WHEN CONVERTING FROM ANY SIMPLEX TO A DUPLEX, THESE CARDS WILL BE REQUIRED (ORDER ONE EACH) FOR THE EXISTING SIMPLEX CEC CABINET FOR DUPLEX CONVERSION.

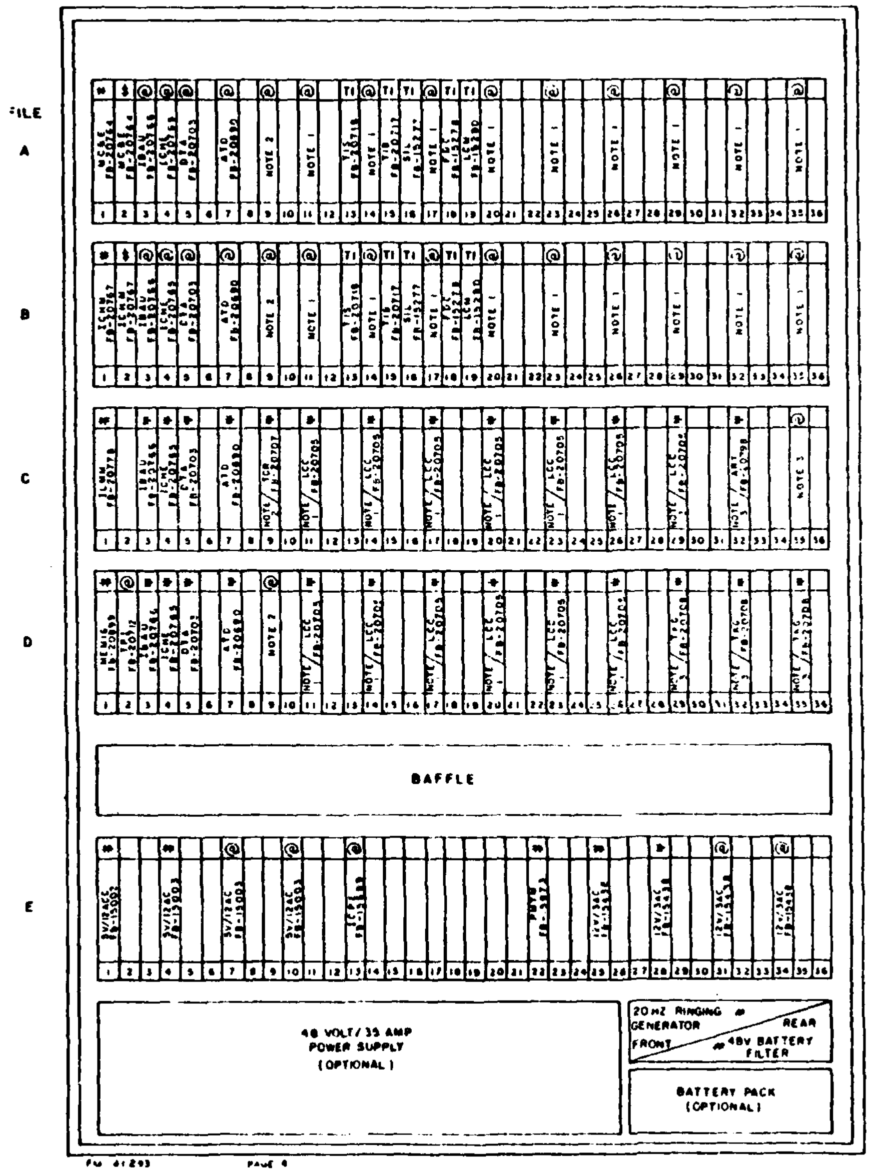
NOTES

1 - OPTIONAL PWC LOCATED IN THESE LOCATIONS ARE:
FB-20920-A (CDTN) REQUIRED FOR CAS OPERATION
OR
FB-20773-A (MDTN) REQUIRED FOR CAMA SENDING.

2 - PWC PART NUMBERS DO NOT SHOW SUFFIX. SEE ORDERING DETAILS CONTAINED HEREIN FOR ASSOCIATED SUFFIX.

Figure 3. Common Equipment Cabinet (CEC) for H/M Option.

GTD-1000 DIGITAL PABX
FULL LINE VERSION
PERIPHERAL EQUIPMENT CABINET (PEC)



CARD GROUP # 1	CARD GROUP # 2	CARD GROUP # 3
FB-20884-A	FB-20772-A	FB-20884-A
FB-20707-A	FB-20707-A	FB-20705-A
FB-20705-A		FB-20708-A
FB-20708-A		FB-20715-A
FB-20715-A		FB-20798-A
FB-20798-A		FB-20775-A
FB-20775-A		FB-15690-A
FB-20772-A		FB-15527-A (FUT.)
FB-15690-A		FB-15528-A
FB-15528-A		FB-15976-A
FB-15976-A		

NOTES:

1. DENOTES UNIVERSALLY WIRED CARD SLOT FOR CARD GROUP # 1.
2. DENOTES UNIVERSALLY WIRED CARD SLOT FOR CARD GROUP # 2.
3. DENOTES UNIVERSALLY WIRED CARD SLOT FOR CARD GROUP # 3.
4. PWC PART NUMBERS DO NOT SHOW SUFFIX. SEE ORDERING DETAILS, CONTAINED HEREIN, FOR ASSOCIATED SUFFIX.
5. USE THIS CABINET IMAGE WHEN DETERMINING CARD LOCATIONS.
6. TRUNKS TO BE USED FOR POWER FAILURE TRANSFER AND 4-WIRE E & M TIE TRUNKS (FUTURE) ARE RESTRICTED TO POSITIONS 29, 32, OR 35 OF FILES C AND D.
7. A MAXIMUM OF 8 TCR'S CAN BE ACCOMMODATED PER PEC

T1 = DENOTES PWC'S REQUIRED FOR T1 TYPE TRUNKING. (NOTE THAT EACH SET OF 5 T1 CARDS IN A FILE [A AND B ONLY] IS CONSIDERED A T1 GROUP. EACH FILE A OR B MUST CONTAIN ALL 5 CARDS IN THE DESIGNATED LOCATIONS FOR THE GROUP TO FUNCTION.)

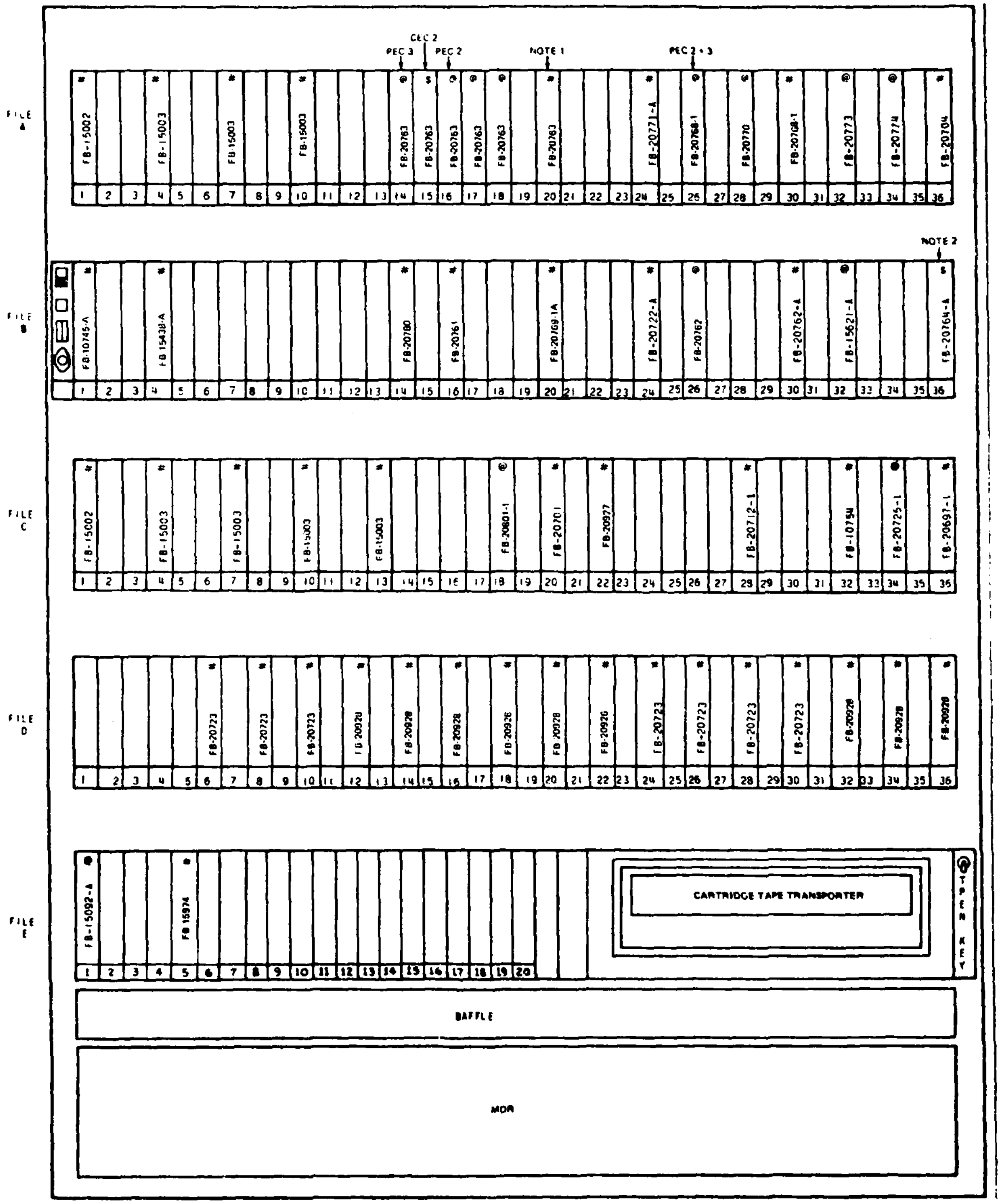
= DENOTES BASIC PACKAGE PWC'S ALWAYS PROVIDED WITH FULL-LINE PERIPHERAL EQUIPMENT CABINET.

* = DENOTES PWC'S ALWAYS PROVIDED WITH FULL-LINE BASIC PACKAGES. WHEN EXPANSION BASIC PACKAGES ARE REQUIRED, THESE CARD POSITIONS MAY BE EQUIPPED AS SHOWN OR WITH CARDS IN THEIR ASSOCIATED CARD GROUP

@ = DENOTES PWC'S REQUIRED FOR OPTIONAL FEATURES AND PROVISION OR EXPANSION OF LINES AND/OR TRUNKS.

$ = DENOTES PWC'S LOCATED IN THESE SLOTS COME EQUIPPED WITH ALL DUPLEX BASIC PACKAGES. WHEN CONVERTING FROM ANY SIMPLEX TO A DUPLEX, THESE CARDS WILL BE REQUIRED (ORDER ONE EACH) FOR THE EXISTING SIMPLEX PEC COPY FOR DUPLEX CONVERSION.

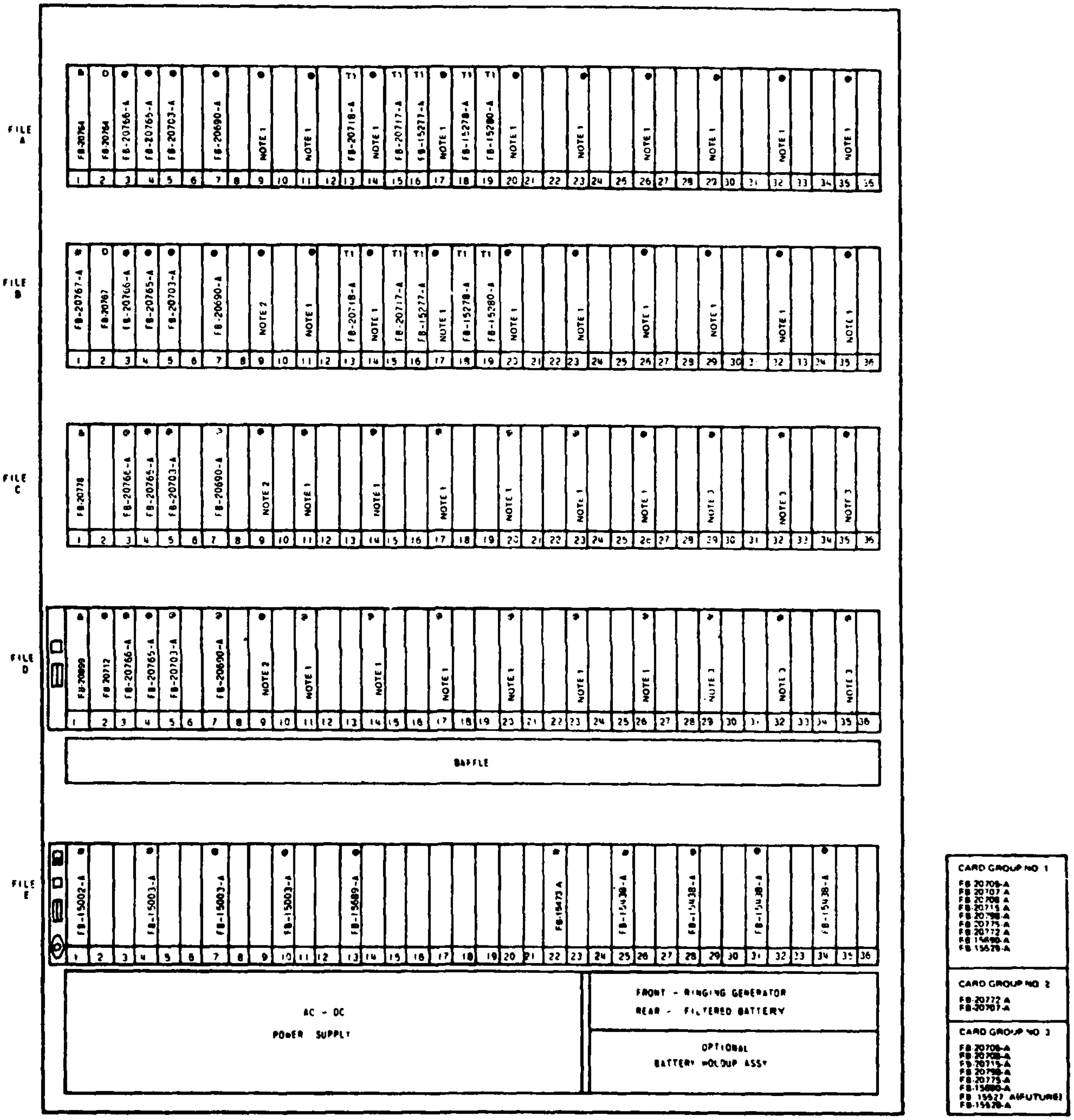
Figure 4. Peripheral Equipment Cabinet (PEC).



LEGEND.
- ✱ INCLUDED IN FULL-LINE COMMON EQUIPMENT CABINET
- ● CARD REQUIRED FOR OPTIONAL FEATURES
- $ REQUIRED FOR DUPLEX OPERATION

NOTES
1. CARDS IN SLOTS 14, 15, 19, AND 28 OF FILE A ARE INCLUDED ONLY WHEN ASSOCIATED PEC'S OR CEC'S ARE USED
2. FOR SIMPLEX CONFIGURATION THIS CARD MUST BE REMOVED.

Figure 5. Common Equipment Cabinet (CEC) for H/M Option and MDR Feature.



LEGEND:
- # INCLUDED IN BASIC PACKAGE
- ● CARD REQUIRED FOR OPTIONAL FEATURES OR LINE AND TRUNK EXPANSION
- ▲ CARD ALWAYS REQUIRED FOR FULL LINE OPERATION
- D CARD REQUIRED FOR DUPLEX OPERATION
- T1 CARD REQUIRED FOR T1 TYPE TRUNKING

NOTE THAT EACH SET OF 5 T1 CARDS IN A FILE (A AND B ONLY) IS CONSIDERED A T1 GROUP. EACH FILE A OR B MUST CONTAIN ALL 5 CARDS IN THE DESIGNATED LOCATIONS FOR THE GROUP TO FUNCTION.

NOTES:
1. DENOTES UNIVERSALLY WIRED CARD SLOT GROUP NO. 1
2. DENOTES UNIVERSALLY WIRED CARD SLOT GROUP NO. 2.
3. DENOTES UNIVERSALLY WIRED CARD SLOT GROUP NO. 3

Figure 6. Peripheral Equipment Cabinet (PEC) for H/M Option.

Earthquake Bracing

2.13 Mounting holes are included in the CEC and PEC to provide earthquake bracing at sites where it may be necessary. Figure 7 provides a top view of the CEC and PEC that shows the eight mounting holes provided with the cabinet. Figure 8 provides a bottom view of the cabinet that shows four of the mounting holes. Earthquake bracing is to be site-engineered by the telephone company in accordance with all local codes.

NOTE: A minimum clearance of 3 feet must be maintained in the front and rear of the cabinets to allow the doors to open.

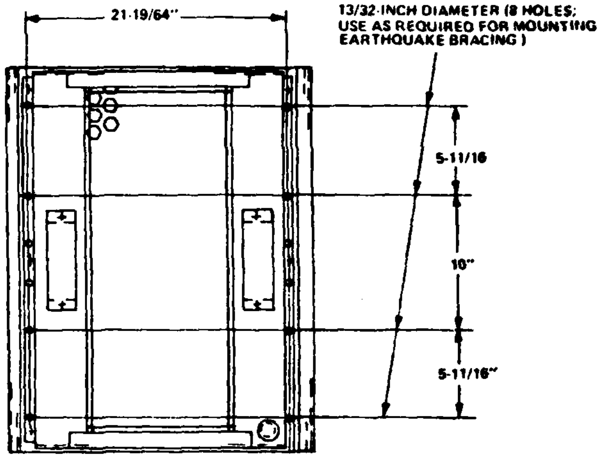


Figure 7. Top View of CEC and PEC, Showing Mounting Holes for Earthquake Bracing.

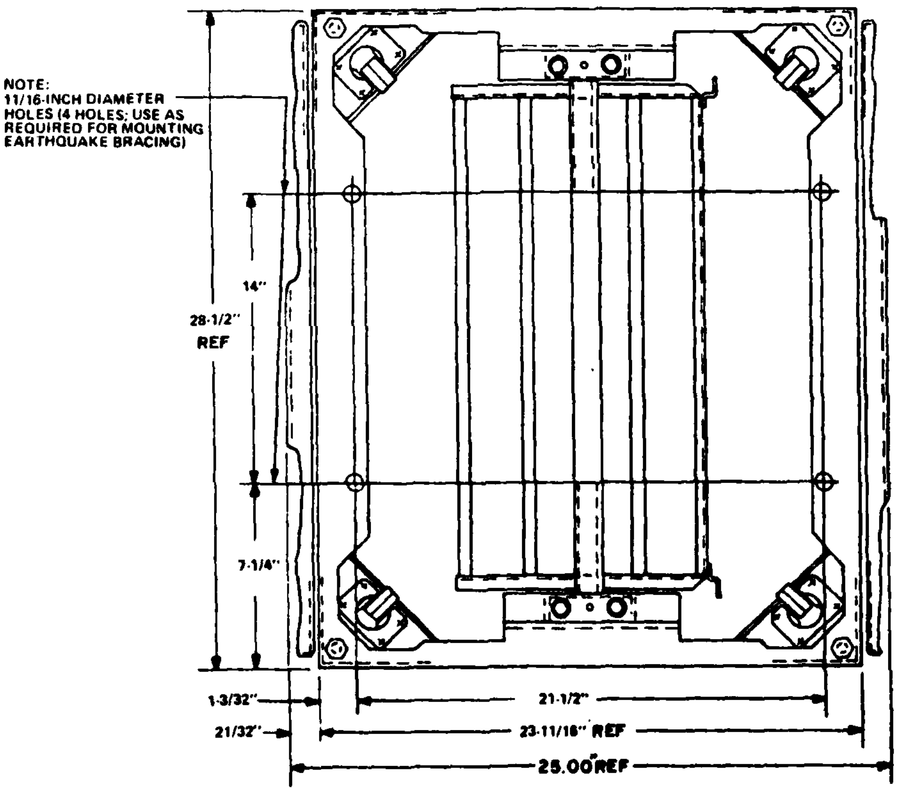


Figure 8. Bottom View of CEC and PEC, Showing Mounting Holes for Earthquake Bracing.

Attendant Equipment

2.14 The attendant equipment provided for the basic system consists of the attendant console (Figure 9), an attendant interface card, and a DTMF receiver card located in a PEC. The attendant console pushbutton arrangements are shown in Figure 10. If the CAS option is implemented, attendant consoles are not required at the branch PABX's. However, they may be provided as an option to handle local PABX traffic.

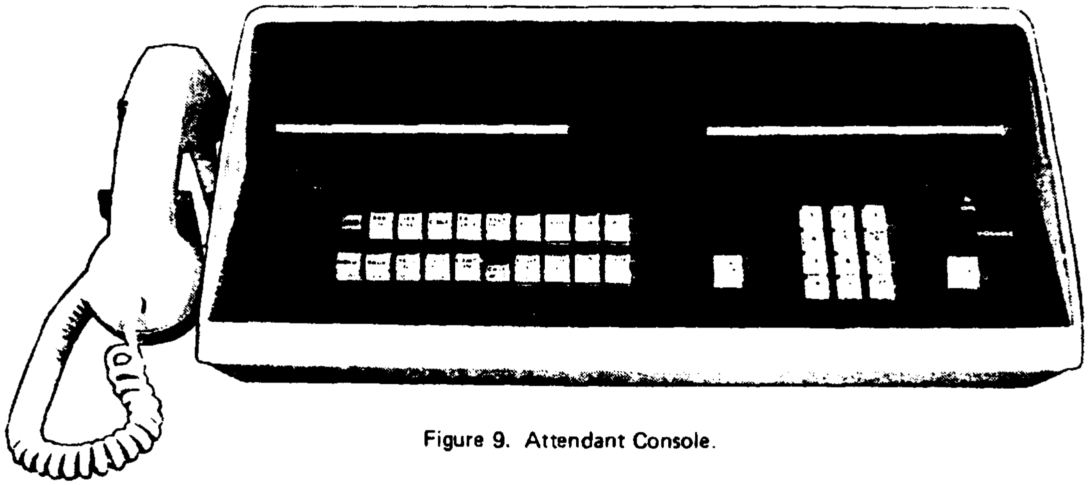


Figure 9. Attendant Console.

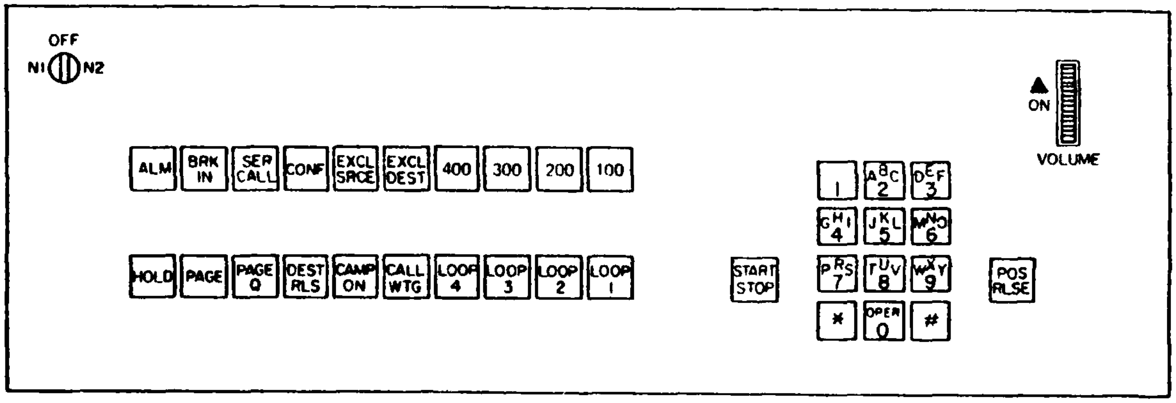
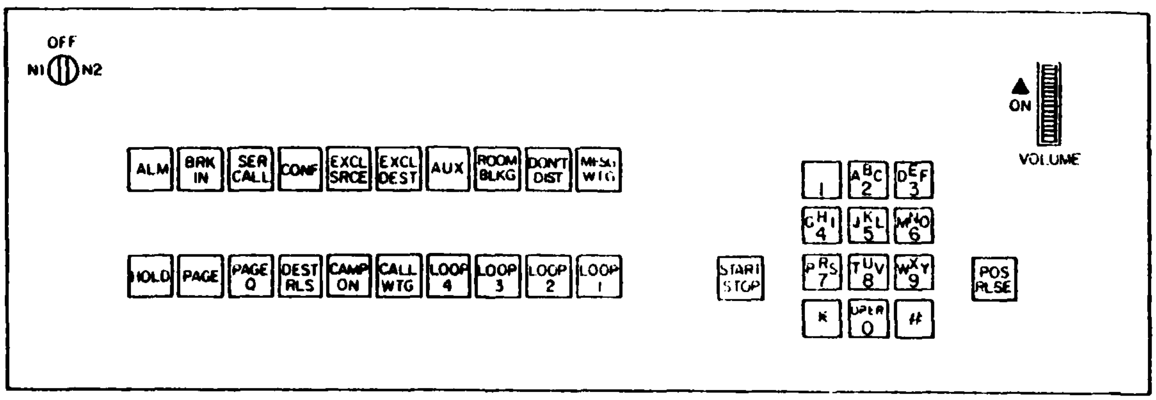


Figure 10a. Attendant Console Pushbuttons Without Health Care/Motel Option.



NOTE:

DESIGNATION TABS "AUX", "ROOM BLK", "DON'T DIST" AND "MESG WTG" ARE LOCATED UNDER "400", "300", "200", AND "100", RESPECTIVELY

Figure 10b. Attendant Console Pushbuttons with Health Care/Motel Option.

2.15 The attendant console provides for day control by the attendant or the selection of various night or unattended control modes. The console is a compact desk-mounted unit with four loops provided. The console contains all controls and indicators needed for complete monitoring and control of the calls (Table 1). Both incoming and outgoing calls are answered, extended, camped on, or released from the console by depressing the appropriate pushbutton.

→ Table 1. Attendant Console Controls and Indicators.

CONTROL OR INDICATOR	TYPE	FUNCTION
N1/OFF/N2	Three-position selector switch	
N1 Position		Automatically transfers any attendant-seeking call to universal night-answer and/or predetermined night-answer mode.
OFF Position		Normal attendant state.
N2 Position		Allows attendant-seeking trunk traffic to be routed to one of four preselected designations (stations or station-hunting groups).
KEYSET	Keyset	A 12-pushbutton DTMF keyset used for extending inward and outward calls.
START/STOP	Pushbutton (nonlocking, lighting)	Indicates start and/or end of dialing to switching equipment.
POS/RLSE	Pushbutton (nonlocking, lighting)	Releases attendant from active loop.
VOLUME	Potentiometer	Regulates volume of signal-chime audio.
BRK IN	Pushbutton (nonlocking, lighting)	Allows attendant to break into nonrestricted busy lines or trunks to verify status and deliver information.
SER CALL	Pushbutton (nonlocking, lighting)	Recalls trunk to attendant after each call completion of a group of calls.
CONF	Pushbutton (nonlocking, lighting)	Adds stations or trunks to conference calls.
EXCL SRCE	Pushbutton (nonlocking, lighting)	Allows attendant private conversation with called party.
EXCL DEST	Pushbutton (nonlocking, lighting)	Allows attendant private conversation with calling party.
100/200/300/400 (Not available with H/M option)	Four Pushbuttons (nonlocking, lighting)	Displays station-busy indication within selected groups.
HOLD	Pushbutton (nonlocking, lighting)	Allows attendant to place incoming calls for service on hold status.

Table 1. Attendant Console Controls and Indicators (Continued).

CONTROL OR INDICATOR	TYPE	FUNCTION
PAGE	Pushbutton (nonlocking, lighting)	Allows the attendant direct access to the paging system as long as the pushbutton is depressed.
PAGE Q (page queue)	Pushbutton (nonlocking, lighting)	Allows the attendant to place incoming page-service calls in queue status.
DEST RLS	Pushbutton (nonlocking, nonlighting)	Allows the attendant to release a station or trunk that elects not to receive a call. Also used in event of depressing the wrong pushbutton.
CAMP-ON	Pushbutton (nonlocking nonlighting)	Allows the attendant to place a waiting call in the camped-on mode to a busy station.
CALL WTG	Pushbutton (nonlocking, lighting)	Indicates to the attendant that an incoming call is waiting for service. Directs call in the call-waiting queue to another attendant (multiple console operation).
LOOP 1 through 4	Pushbuttons (nonlocking, lighting)	Provides the attendant with an indication of a loop having a call that requires service.
Additional Controls and Indicators Included With the H/M Option		
MESG WTG	Pushbutton (nonlocking, lighting)	Allows the attendant to initiate or cancel the message waiting feature and also provides visual indication of the feature implementation.
DON'T DIST	Pushbutton (nonlocking, lighting)	Allows the attendant to initiate or cancel the do-not-disturb feature. Also provides visual indication of the feature implementation.
ROOM BLKG	Pushbutton (nonlocking, lighting)	Allows the attendant to initiate or cancel room-to-room blocking. When blocking is in effect, the pushbutton is lit.
AUX	Pushbutton (nonlocking, lighting)	No function at this time.

2.16 An attendant console type-of-call display (Table 2) is provided by a lighted indicator. The type of call remains indicated as long as the attendant is servicing the call. The attendant type-of-call display also lights on attendant recalls.

2.17 Attendant console status indicator lamps (Table 3) provide call status information to the attendant. These indicators in the full-line version include a 100-LED station-busy display, a trunk-and-station-number display, a class-of-service display, a trunk-status display, and a system-fault alarm. Indicators for the H/M option include a trunk-and-station-number display, a class-of-service display, a room number feature display, and a system-fault alarm.

→ Table 2. Attendant Console Type-of-Call Display.

DISPLAY	FUNCTION
LOCAL	Indicates an incoming CO call.
INFO	Indicates that a station desires assistance of the attendant and has dialed 0 (when flashing indicates automatic recall of a call on hold).
SERIES	Indicates that an incoming series call has been completed. On an outgoing series call (attendant-controlled WATS), is displayed when the PABX station has disconnected and the attendant has answered the series indicator.
NO ANSWER	Indicates that a station waiting for a connection and has completed the delay or camp-on interval.
STILL BUSY	Indicates that a camp-on call has remained for a predetermined period and the call is diverted to the attendant or the attendant has transferred a call to a busy station and then released from it.
TIE LINE	Indicates an incoming tie-line call.
FX	Indicates an incoming call from a foreign exchange.
INTCPT	Indicates that a station has attempted a call not in its class-of-service and has been routed to the attendant.
L.D. RES	Indicates an attempted toll call from a restricted station and diversion to the attendant.
TRNSF	Indicates an attendant recall, when a station user has momentarily depressed the hookswitch and dialed 0.
WATS	Indicates an incoming WATS trunk.
SPL1	Flashes at 60 ipm for the do-not-disturb feature and is lit for off-hook routing.

→ Table 3. Attendant Console Status Indicator Lamps.

LAMP	FUNCTION
STATION BUSY (Not available with H/M option)	100-LED lamp display. LED shows status of four groups, each containing 100 stations.
MESSAGE TYPE	Indicates type of call as described in Table 3.
CLASS OF SERVICE	Indicates the class of service associated with the calling party.
TRUNK STATUS	Indicates trunk-group status of 20 trunk groups and a maximum of 8 restricted trunk groups.
TRUNK AND STATION	Indicates the number of the calling station (internal) or the number of the calling trunk.
ALARM	Indicates a system malfunction.
FEATURE TYPE (Only available with H/M option)	Indicates the status of the room number feature.

2.18 The attendant interface card in the PEC provides data and transmission links between the attendant console and the system. The card contains the relays for the universal night-answer mode and the contact for the failure-alarm indicator.

2.19 Headset/handset jacks are provided on both sides of the console. When the headset/handset is removed from the jack, the console is considered busy and the system is placed in the N1 universal night-answer mode of operation, or the call is diverted to a second console. The console connection to the system is via a multiplexed data link. If the data link fails, the system transfers to the N1 universal night-answer mode.

2.20 All attendant-seeking traffic is queued on a FIFO basis. Only one new call is presented to the attendant at any given time. The attendant may release from the active loop by depressing the POS RLSE, HOLD, or CAMP ON pushbutton, or another LOOP pushbutton. Upon release, the next call in the call-waiting queue is presented to the attendant.

3. SYSTEM ORGANIZATION

3.01 The system hardware, as shown by the block diagram in Figure 11, consists of the following:

(a) Central Processor Unit (CPU).
(b) Time-switch network.
(c) Peripheral circuitry.
(d) Miscellaneous circuitry.
(e) Pulse Code Modulation (PCM)

3.02 The network and the CPU control the processing of the data exchange within the system hardware. The network is a completely solid-state, digital time switch that temporarily stores and transmits PCM samples to the appropriate terminations. The timed switching is done at speeds compatible with present-day PCM carrier equipment, thus permitting implementation of two T1-type span lines per PEC (24 channels each) directly into the system.

4. CENTRAL PROCESSOR UNIT

Description

4.01 The CPU contains all the decision-making logic in the system and the network call setup. It scans all terminations for supervisory and signaling information. The basic clock rate is 500 nanoseconds per machine cycle, with a minimum of four machine cycles required per instruction. The CPU interfaces with the other subsystems through the following circuits:

(a) Sixteen-bit address bus.
(b) Eight-bit data-in bus.
(c) Eight-bit data-out bus.
(d) Control signal leads.

4.02 The CPU consists of the following circuits:

(a) Enhanced Intermediate Central Module Microprocessor (EICMM).
(b) Central Processor Unit (PCU) Interrupt (CPI).
(c) Expanded Memory Paging Buffer Controller (EMPBC).
(d) An instruction unit 0 memory (MEM) consisting of 64 K eight-bit words (K=1,024) and an instruction unit 1 consisting of 40 K eight-bit words expandable to 64K.
(e) A data unit 0 memory (MEM) consisting of 56 K eight-bit words and a data unit 1 consisting of 24 K eight-bit words expandable to 56 K.
(f) Microprocessor Computer Bus Interface (MCBI).
(g) Microprocessor Computer Bus Extender (MCBE).

Features and Services

4.03 The features and services provided by each major circuit of the CPU are described in the following paragraphs.

4.04 The EICMM is a general-purpose central processor designed around a microprocessor chip. In addition to the microprocessor and associated interface logic, it contains the following items:

(a) Central Processor Unit (CPU) clock.
(b) Read-Only Memory (ROM), 512-word-by-8-bit.
(c) Eight-level priority-interrupt structure.
(d) Status and control.

4.05 The CPI consists of two independent circuits: the processor interrupt and the processor reset. The CPI provides both an interrupt signal and all reset signals required by the CPU. The interrupt circuit generates an interrupt to the EICMM once every 10 milliseconds. The EICMM can enable, disable, and clear the interrupt circuit under software control. This interrupt called the 10-millisecond interrupt, is used by the EICMM as a timebase for scheduling of software tasks that must be executed on a periodic basis. The processor reset circuit generates a 1.5-microsecond reset signal to the EICMM when any one of the following events occur:

(a) System power-up.
(b) Manual system reset.
(c) Time-out of a 500-millisecond watchdog timer. (The watchdog timer is enabled and cleared under software control. The EICMM must clear the watchdog timer at least once every 500 milliseconds, ensuring that the EICMM does not become trapped in an endless software loop).

4.06 The EMPBC performs a memory paging function, allowing separate pages of data and instruction memory. During program execution, instructions that will be executed by the EICMM are read from the instruction (or

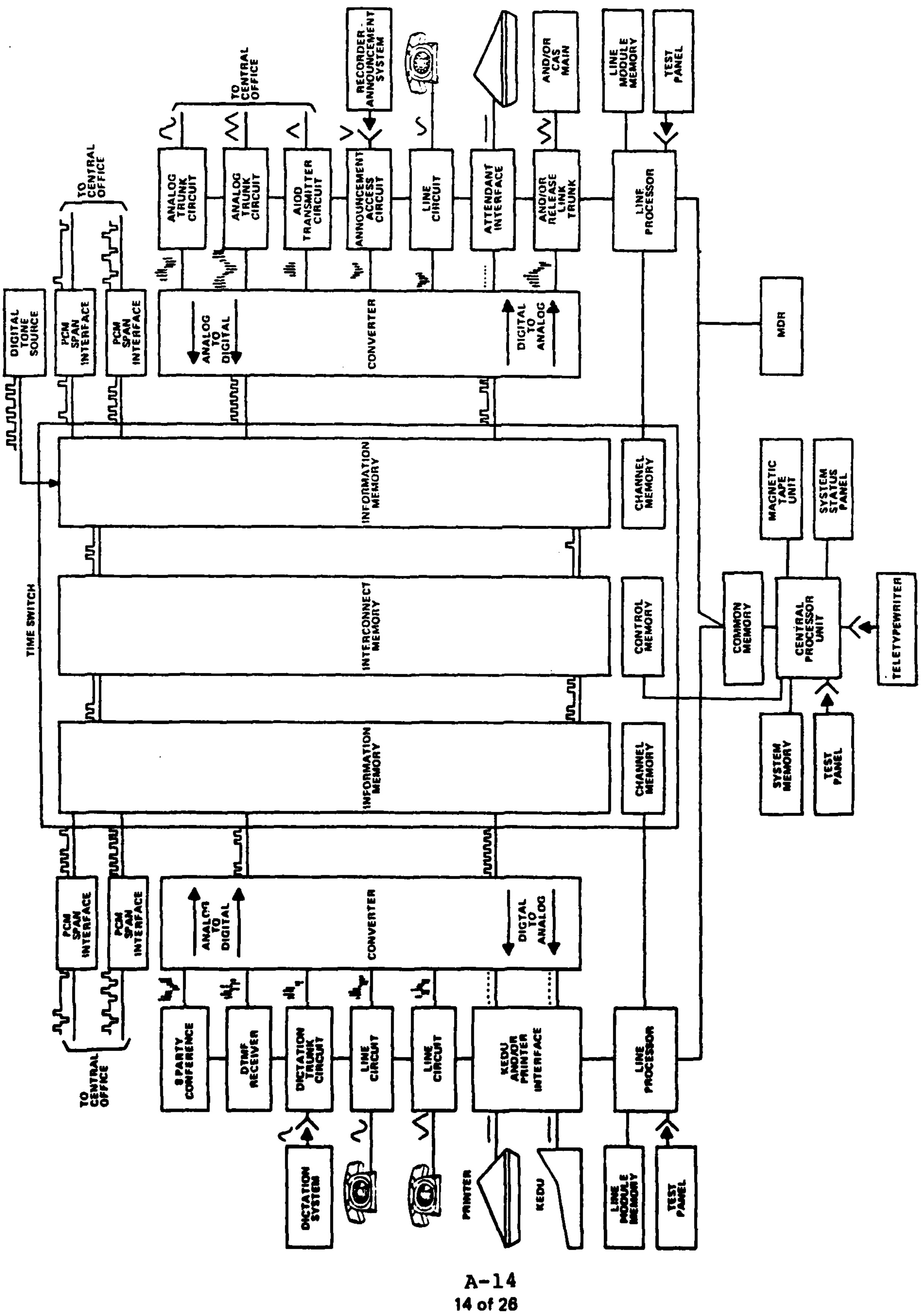


Figure 11. System Block Diagram.

generic) page. However, when references must be made to the data page, the EMPBC is automatically switched to allow data to be read from, or written to, the data page. Also, the paging mode may be set under software control to allow reading and/or writing of either data or instructions on the desired page. The EMPBC provides buffering of all memor-related signals from the EICMM.

4.07 The MEM serves as a random-access memory for the generic, data-base, and scratch-pad software subprograms. The memory consists of four pages: two instruction pages and two data pages. Each instruction page, consisting of eight MEM cards, contains the generic software. Each data page, consisting of eight MEM cards, contains the data-base and scratch-pad areas. In addition, there is another 4K of memory area that is accessed by all processors in the system. This common memory area is accessed as though it were data-page memory and is used as a means of allowing the EICMM to communicate with the other processors in the system.

4.08 The MCBI provides the connection from the CEC to the MCBE located in the PEC, or in the CEC in a duplex system. The MCBI's are arranged in a priority system, which ensures that only one processor at a time may access common memory.

4.09 The MCBE allows its processor to request common memory access to the CEC via its MCBI in the CEC. When access is granted, the processor may read or write in common memory. If access is not granted within 16 microseconds of the request, a time-out interrupt is generated to the requesting processor.

5. TIME-SWITCH NETWORK

Description

5.01 The time-switch network provides the digital network and control circuitry that is the point where the actual transfer of the encoded voice communications between the calling and called parties takes place. The time-switch network consists of the following circuits:

(a) Intermediate Network Clock (INCK).
(b) Intermediate Network Control Memory (INCM).
(c) Intermediate Channel Memory (ICHM).
(d) Digital Tone (DTN).
(e) Intermediate Network Information Memory (INIM).
(f) Intermediate Network Interconnect Card (INIC).

Features and Services

5.02 The features and services provided by each major circuit of the network are described in the following paragraphs.

5.03 The INCK provides the basic timing pulses to the network, and the PCM peripheral circuitry controls the network functions. A 49.408-MHz clock generates the basic timing pulses that are channeled to the time-slot counter, which forms the following pulse trains:

(a) Time-slot pulse trains.
(b) Channel pulse trains.
(c) Frame pulse trains.

5.04 The INCM consists of the following items:

(a) Central Processor Unit (CPU) data.
(b) Central Processor Unit (CPU) address.
(c) Central Processor Unit (CPU) control leads.
(d) Read data.
(e) Steering.
(f) Two 192-word-by-8-bit memories.

5.05 The two 192-word-by-8-bit INCM's are addressable by the CPU. Each word in the INCM relates to corresponding words in the channel and information memories. An address in the INCM identifies a channel that is to send information to the current channel being scanned.

5.06 By means of an instant-speaker algorithm, the second INCM permits three-way conferencing by using only the basic time switch. When a line is scanned, the time switch obtains PCM samples from the two addresses contained in control memory A and control memory B. The absolute values of the samples are compared and the sample with the largest value is sent to the line being scanned.

5.07 The ICHM is a 96-word-by-8-bit read-write memory that is sequentially read under control of the early counter at the beginning of each time slot. It can be randomly read or written on request from the CPU only during the last quarter of any time slot. The sequential operation of the ICHM is slaved to the clock, to the PCM interface, and to the control memory INCM controls for the random operation originated by the CPU. The function of the ICHM is to read the word indicated by the early counter and store it in the buffer.

5.08 The ICHM provides the interface to the lower-level common equipment represented by the DTA converters, the ATD converters, and the ICHE circuits, and DTN's. The ICHM card also contains the channel-shift register and the early counter.

5.09 The DTN generates and encodes tones, such as dial tone, ringback tone, busy tone, etc, into their digital PCM representation. These PCM samples are stored in an ROM and are accessed by the time switch and sent to the appropriate line, trunk, etc, under the direction of the CPU. The tone outputs coming from the ROM look identical to encoded tone samples and, after conversion to analog, sound like conventional tones.

5.10 The INIM contains the actual time switch where PCM voice samples are stored. The INIM also contains three-way conferencing and digital pads. One INIM serves two PEC's.

5.11 In a system using more than two PEC's, the INIC is the interface between the INIM serving PEC cabinets 0 and 1 and the INIM serving PEC cabinets 2 and 3. The INIC contains the exact information stored in both INIM's; thus, each INIM can read the contents of the other INIM via the INIC.

6. PERIPHERAL CIRCUITRY

Description

6.01 Peripheral circuitry provides the interface between customer stations, central offices, other PABX's, and the system switching mechanism. This circuitry receives and sends the same analog, signaling, and supervisory information as conventional PABX's. A stored-program approach has resulted in the elimination of substantial logic in the trunk circuitry, permitting extensive size reduction over conventional circuitry. The peripheral circuitry consists of the following circuits:

(a) Line Circuit (LCC).
(b) Loop or Ground Start Trunk Circuit (TKG).
(c) E&M Tie-Trunk Circuit (TKT).
(d) DTMF Receiver Circuit (TCR).
(e) Bus Amplifier Circuit (IBAU).
(f) Channel Enable Circuit (ICHE).
(g) Digital-to-Analog Converter Circuit (DTA).
(h) Analog-to-Digital Converter Circuit (ATD).
(i) Release Link Trunk (RLT) circuit, if the CAS option is implemented.
(j) Direct Inward Dialing (DID) trunk.
(k) Conference card circuit (ICONF).
(l) Attendant interface card (ART).
(m) Dual Modem and Loop Interface (DMLI) card.
(n) Dictation and paging card (IDPE).
(o) An AIOD card.

Features and Services

6.02 The features and services provided by each major circuit of the peripheral circuitry are described in the following paragraphs.

6.03 The LCC provides the analog interface between a battery-fed telephone and the DTA and ATD circuits of the system. In addition, the LCC provides the following operations:

(a) Detects on-hook and off-hook conditions.
(b) Routes battery or ringing current to the station apparatus.
(c) Disconnects ringing current when the off-hook condition is detected (ring trip).
(d) Outputs line status to the CPU when interrogated.
(e) Operates or releases the ringing relay as dictated by the CPU.

6.04 The TKG is a two-way trunk. Outgoing, the TKG has line-circuit access at the central office; incoming from the central office, it has connector access.

6.05 The TKG contains two control points and four sense points for use by the CPU to check line conditions and present signals to the T&R leads. On incoming calls, the TKG, in conjunction with the attendant console, provides for termination (answering and extension to station users). On outgoing calls, the system connects the audio portion of the local line in the system to the line access in the central office via the switching network. The common control receives signaling from the local line and forwards it through the TKG to the central office.

6.06 The TKT is a two-way E&M trunk to another PABX. When used in an outgoing mode, the CPU scans for an idle TKT as a result of a dialed access code. When an idle TKT is found, the CPU sends a signal to the distant-office trunk. The common control performs DP signaling by forwarding signals from the line to the TKT.

6.07 On incoming calls from a distant office, the CPU checks the E-lead sense point to see if a change of state has occurred since the last scan. A software register collects the digits as they appear on the E-lead sense point, and the CPU makes the desired connection to a terminal when sufficient digits have been registered.

6.08 The DTMF receiver receives and decodes the 2-out-of-8 DTMF tones from a station's line and outputs the information in hexadecimal code.

6.09 The IBAU performs the following basic functions:

(a) Generates circuit-enable signals for the trunk, line, and special receiver circuits.
(b) Selects one of four trunk circuits from each trunk circuit card.
(c) Buffers the low-level data from the trunk, line, and special receiver circuits to the bus interface circuit.
(d) Buffers the bus interface circuit high-level data to be written into the trunk, line, and special receiver circuits.
(e) Transfers the high-level data to the next bus interface circuit in the next frame.
(f) Buffers the read and write signals coming from the bus interface circuit.

6.10 The ICHE provides for the allocation of 24 channel-enable pulses to a group of equipment by receiving and decoding an eight-bit equipment-identity number and distributing the resulting transmit pulse and/or enable pulse to the assigned equipment.

6.11 The DTA converts the eight-bit-parallel PCM input received from the information memory to a Pulse-Amplitude-Modulated (PAM) signal that it transmits to the line and trunk circuits. Each line and trunk circuit samples and

filters the PAM bus at a time determined by the ICHE, thereby transforming the PAM sample to an analog-output signal.

6.12 The DTA consists of the following circuitry:

(a) Pulse Code Modulation (PCM) receive.
(b) Ladder network.
(c) Pulse Amplitude Modulation (PAM) generator.

6.13 The ATD provides the analog and digital circuitry necessary to convert the PAM signals from the line, trunk, and DTMF receivers to an eight-bit binary code.

6.14 Release Link Trunks (RLT's) are used when the CAS option is implemented. The RLT's provide one-way connections from the branch PABX's to the CAS main location for sending supervisory and information signals. An RLT can only be seized by its associated branch PABX. An RLT cannot be seized by equipment at the main location. An RLT is connected to the main location by tip and ring leads. During the idle state, the tip and ring leads are open. When an RLT is seized by the branch PABX, the RLT simultaneously provides ground on the tip lead, and PABX steady ringing on the ring side. After answer and until disconnect, the RLT provides ground through a maximum of 425 ohms resistance on the tip lead, and −48 Vdc through a maximum of 425 ohms resistance on the ring lead.

6.15 The DID trunks provide one-way incoming loop seizure. A DID trunk from an SxS office appears to the SxS office as the final selectors and connectors. The system must be arranged to receive up to seven digits within 65 ms of seizure. Upon completion of dialing, the system rings the station and returns an audible ringback tone. On answer, the system returns answer supervision. When CO's are arranged for sender outpulsing on DID calls, delay-dial or wink-start supervision is required.

6.16 An optional conference circuit added to the PEC permits conference calls between up to eight parties, lines, or trunks. It is recommended that no more than two trunks be connected to the conference bridge to maintain satisfactory transmission quality.

6.17 The ART provides an audio interface port between the attendant console and the remainder of the system. It also provides the universal night-answer feature for four zones and the power alarm sense/control point. It also acts as the data link between the multiplex console and the system.

6.18 The DMLI card for the H/M option provides the interface between the system and the KEDU's and/or printers. Two KEDU's or two printers or one KEDU and one printer are allowed per card.

6.19 The IDPE contains the following circuitry:

(a) Four paging adapter relays to interface the two-wire transmission circuit to customer-owned paging equipment.
(b) A paging adapter to provide the option of connecting a music system through the adapter relay to the public address system when paging is not in use.

6.20 An optional AIOD circuit located in the PEC provides the identity of the calling station on all outgoing calls routed via the associated central office trunks.

7. MISCELLANEOUS CIRCUITRY

Description

7.01 The miscellaneous circuitry consists of the following items:

(a) Data cartridge drive unit.
(b) Power Monitor Transfer and Low Voltage Alarm (PMTV) card.
(c) Power Monitor Transfer and Call Message Waiting Control (PMTM) card.
(d) Maintenance-Test-Panel Interface (TPI) card.
(e) Cartridge-Read Interface (CRCI) card.
(f) Cartridge-Write Interface (CWCI) card.
(g) Cartridge Direct Memory Access (CDMAC) card.
(h) Serial Device Control (SDC) card.
(i) System Status, Configuration, and Control (ISCC) card.

Features and Services

7.02 The features and services provided by each major circuit of the miscellaneous circuitry are described in the following paragraphs.

7.03 A data cartridge drive is incorporated as a permanent part of the system. It is used to load the Random-Access Memory (RAM) and to write onto magnetic tape any changes made on site to the data base and generic program. The tape generated on site is then used as the backup for the system software when a reload of the software is required.

7.04 The data cartridge drive provides the following features:

(a) Phase-encoded, 1,600-bit-per-inch (bpi) recording density.
(b) Read/write speed of 30 inches per second (ips).
(c) Four-track read-while-write tape head.
(d) DC tachometer for accurate speed control.
(e) Transistor-Transistor Logic (TTL) interface compatibility.
(f) Single-motor bidirectional operation.

7.05 The PMTV card is used in the CEC. This circuit performs the following functions:

(a) Monitors and displays the status of all power supplies and fuses within the cabinet.
(b) Provides four transfer relays, under both manual and/or software control.
(c) Provides a start signal under switch and voltage monitor control.
(d) Provides an alarm interface with the system software and a contact set for remote alarm control.
(e) Provides two toggle switches for manual control as follows:
 (1) Momentary switch for system reset.
 (2) Transfer switch for releasing normally operated power-failure relays.

7.06 The PMTM card is used in the PEC. This circuit performs the following functions:

(a) Monitors and displays the status of all power supplies and fuses within the cabinet.
(b) Provides for four power-failure transmission loops that provide a guarded restoration of the power-failure relays. The power transfer relays are software or manually controlled.
(c) Provides a central processor reset when power is activated and under manual control.
(d) Provides an alarm interface with the system software and a contact set for remote alarm control.
(e) Provides call message waiting potential and control logic via system CPU.

7.07 The TPI provides the interface between the portable maintenance test set and the system. The TPI operates by providing input/output signals to the system in response to control settings on the test set. It also provides signal levels to the test set based on signals generated by the TPI.

7.08 The CRCI interfaces the CPU to the magnetic tape cartridge transport. The CRCI conta[illegible] all the circuitry necessary for reading the motion-control status and the data status from the tape.

[illegible]9 [illegible] CWCI [illegible] a partitioned logic card designed to [illegible] that are compatible with the proposed [illegible] Standards Institute (ANSI) Standard [illegible] magnetic tape cartridges.

[illegible] with the CRCI and [illegible] operations to be [illegible]

[illegible]

7.12 The ISCC circuit provides a three-position toggle switch to set the CEC definition as CEC0, CEC1, or out of service. This switch is read only on power-up or system reload. The ISCC circuit also provides nine system status lamp indications, as follows:

(a) The ON-LINE lamp, when lit, indicates that its respective CEC is on-line. It is controlled by the software address and not by switch position.
(b) The CEC OK lamp, when lit, indicates that any self-diagnostic subroutines detect no malfunction.
(c) The OTHER CEC OK lamp, when lit, indicates that communication from the other CEC is satisfactory.
(d) The PECO-3 OK lamps, when lit, indicate that the particular PEC is communicating properly with the CEC.
(e) The ALARM lamp, when lit, indicates a fault condition. This lamp is accessed both by software address and hardware input. A spare lamp is provided for possible future use.

The CEC OK, OTHER CEC OK, and PECO-3 lamps, when flashing, indicate that a communication problem exists but the respective unit is operating properly. The ISCC circuit also provides for the gating of various interrupts and control signals to the peripheral equipment and other CEC cabinets when the system is in the multiprocessor configuration.

7.13 The ISCC circuit provides decode and control points for loop clock synchronization when used with the digital trunk interface cards. Network clock failure is decoded by this circuit. The ISCC circuit provides sense points for low voltage and power failure.

8. DIGITAL TRUNK INTERFACE

Description

8.01 The T1-type digital trunk interface between the Time-Division Multiplexing (TDM) span and the system digital network, performs the following functions:

(a) Synchronization of PCM channels.
(b) A buffer between incoming and outgoing PCM signals.
(c) Conversion of the PCM signal from unipolar to bipolar and vice versa.
(d) Addition and removal of signaling and supervisory information from the PCM signal.
(e) The decoding of channel A and B signaling.

8.02 The T1 type digital trunk interface is accomplished by replacing three analog trunk circuit printed wiring cards with the following digital trunk cards:

(a) Span Interface (SIL)
(b) Frame Detector (FDC)
 [illegible] (LCM)
(c) [illegible] (T1R)
 [illegible] (T1S)

Features and Services

8.03 The features and services provided by the digital trunk circuit cards are described in the following paragraphs.

8.04 The SIL receives and terminates the incoming bipolar signal, converts it to the incoming unipolar bit stream, and extracts the clock frequency from the bipolar stream. It converts the outgoing unipolar stream into a bipolar signal that is compatible with the T1-type line. The span interface card also provides the looping ability to test the digital cards for synchronization.

8.05 The FDC performs the following functions:

(a) Monitors for errors in synchronization patterns.
(b) Provides an alarm signal to the T1-type supervisory card when three or more synchronization errors are detected out of five incoming bits examined.
(c) Signals the span interface card that a new frame of voice samples is about to arrive.
(d) Provides the supervisory frame signal that decodes channel A and B signaling.

8.06 The LCM card converts the unipolar bit stream received from the span interface card to an eight-bit voice sample that is forwarded to the T1-type buffer card in parallel format. The LCM further provides buffering to compensate for the variations in temperature of the span line.

8.07 The T1B provides a buffer between the incoming PCM signal from the line compensator card and the outgoing PCM signal from the digital network. It synchronizes and aligns the 24 PCM channels between the digital network and the digital trunk interface cards.

8.08 The T1S provides a strapping field to select a Foreign Exchange (FX) trunk interface or a standard off-hook/on-hook interface such as E&M channels. This strapping field provides the means of selecting either the D2-type or D3-type signaling format. The card provides the supervisory signaling interface between the system and the T1-type interface cards. This card contains the following lamps and switches:

(a) A local alarm (LOC) lamp that lights when the framing of the incoming bipolar signal is lost.
(b) A remote alarm (REM) lamp that lights when the bit position two of the incoming bipolar stream is inhibited for a period of 1.32 to 1.44 seconds.
(c) A system alarm (SYS) lamp that lights when any alarm condition exists, including when the system is fully synchronized but in a loop mode.
(d) A remote power failure alarm (RPF) lamp that lights when a power failure occurs in the office terminating shelf.
(e) An alarm cutoff (ACO) lamp that lights when the alarm cutoff switch is activated.
(f) A two-position alarm cutoff switch that is activated in the up position.
(g) A loop switch (LOP) lamp that lights when the loop test switch is activated.
(h) A two-position loop switch that is activated in the up position. This switch activates the loop test.

9. SOFTWARE

9.01 All major call-processing functions are performed by the system software. Software is divided into two parts: one part resides in the central processor and the other part resides in the peripheral processor.

9.02 The software residing in the central processor is divided into the following subprograms:

(a) Executive.
(b) Common memory.
(c) Call control.
(d) Attendant console control.
(e) Digit analysis.
(f) Common Equipment Cabinet (CEC) administration.
(g) Common Equipment Cabinet (CEC) maintenance.
(h) Peripheral Equipment Cabinet (PEC) control.
(i) Hardware interface.
(j) Central processor interface.
(k) Line control.
(l) Station features.
(m) Trunk control.
(n) System features.
(o) Centralized Attendant Service (CAS) branch.

Executive Subprogram

9.03 The executive subprogram maintains real-time control of the system environment and is responsible for performing the following operations:

(a) Job scheduling.
(b) Monitor and control of system operations.
(c) Allocation of common system resources.

9.04 In addition to the above routines, the operating subprogram provides routines within the supervisory program that accomplish process-report real-time services for the application programs.

9.05 The executive subprogram is divided into the following two schedulers:

(a) Foreground scheduler.
(b) Background scheduler.

9.06 The foreground scheduler consists of a dispatcher that periodically schedules the following routines:

(a) Peripheral processors foreground scheduler activation.
(b) System real-time clock update.
(c) Outpulsing DTMF and MF tone.
(d) Sending CAS audible tone identification.

9.07 The background scheduler consists of a dispatcher that schedules routines that do the following:

(a) Check for 100-ms, 500-ms, 1-second, and 1-minute scheduled jobs to be done.
(b) Process events that are reported from the peripheral processors. These events include line- and trunk-status changes, such as on-hook and off-hook conditions and digit presentation, attendant-console-pushbutton actions, and system maintenance test results.
(c) Process information obtained by the foreground scheduler.

Common Memory Subprogram

9.08 The common memory subprogram (also called peripheral processor interface subprogram) links the central processor to the peripheral processors through a common memory. This subprogram performs the following operations:

(a) Passes the action directive to the peripheral processors.
(b) Reports the events from each peripheral processor and passes them on to the proper processing application program.

Call-Control Subprogram

9.09 The call-control subprogram processes the events reported by the digit-outpulsing and trunk-scan routines and the console-handler and digit-analysis subprograms, and allocates the following system resources when required:

(a) Call stores.
(b) Register-senders (digit stores).

9.10 Call Stores. The status of all calls in progress is maintained in allocated blocks of memory known as call stores. These call stores contain the current status of specific calls. A call store is allocated at the time a line or trunk seizure is processed. The call store retains this information throughout the duration of the call.

9.11 Register-Senders (Digit Stores). In addition to the call stores, there are register-sender (digit store) memory blocks dynamically allocated upon processing of a line seizure that are linked to the associated call store during digit accumulation and outpulsing. The register-sender (digit store) memory contains the control words required to accumulate, store, and outpulse a maximum of 16 digits.

Attendant Console Control Subprogram

9.12 The attendant-console-control subprogram performs the following operations as required by the background scheduler:

(a) Analyzes the status of the attendant console pushbuttons as reported by the peripheral-processors' status-handler-routine.
(b) Causes the appropriate lamps on the attendant console to light.
(c) Reports the status of the attendant-console pushbuttons to the call-processing subprograms for further disposition.

9.13 As an example, depressing one of the LOOP pushbuttons and the START/STOP pushbutton on the attendant console causes the remote-console-handler subprogram to light the LOOP, POS/RLSE, and START/STOP pushbuttons and report the event to the call-processing subprogram. The call-processing subprogram then allocates a call store, a register-sender, and a DTMF receiver to that loop.

Digit-Analysis Subprogram

9.14 The digit-analysis subprogram is activated by the presence of a digit and performs the following operations:

(a) Analyzes each received digit.
(b) Checks system features against the codes dialed.
(c) Checks to ensure that a connection can be made to a terminator.
(d) Reports the type of equipment to which the call has been terminated.

9.15 On station-to-trunk calls, the digit-analysis subprogram monitors the outpulsed digits and performs the appropriate digit-absorption and code-screening functions.

Common Equipment Cabinet Administration Subprogram

9.16 The CEC-administration subprogram consists of the following routines:

(a) Memory-reload routines.
(b) Write-to-tape routines.
(c) Traffic-data-verification routines.
(d) Input/output routines.

9.17 The memory-reload routine in the central processor reloads information into the system's generic and data base memories when the watchdog timer times out or on a return from a power failure. The routine is resident in the ROM and, when activated, reloads all the software data base. The memory reload function in the peripheral processor is controlled by the on-line central processor.

9.18 The write-to-tape routine, when activated, performs the following functions:

(a) Writes and reads a test block of data to verify proper system operation.
(b) Outputs the data base and generic programs to magnetic tape in a multitrack ANSI format (one track for central processor generic, one track for peripheral processor generic, one track for office-dependent data base).
(c) Duplicates the output of the data blocks onto magnetic tape to ensure against faulty magnetic tape areas.
(d) Advances the magnetic tape to the End-of-Tape (EOT) marker and primes the machine to perform the reload function when required.

9.19 The traffic-data-verification routine, when activated, performs the following functions:

(a) Collects event and usage data.
(b) Automatically outputs the collected data, if desired.
(c) Provides an interface for remote data-collection systems.

9.20 The input/output routine, when activated, performs the following functions:

(a) Provides an interface for the digital system test set.
(b) Provides an interface for the TTY.
(c) Provides local/remote memory read/write capability.

Common Equipment Cabinet Maintenance Subprogram

9.21 The CEC-maintenance subprogram consists of the following routines:

(a) Fault-detecting test routines.
(b) Fault-analysis and fault-report routines.
(c) System-reconfiguration and call-recovery routines.

9.22 Fault-detecting-test routines are run continuously to detect the following faults:

(a) Memory faults.
(b) Network faults.
(c) Inter-processor communication faults (central processor-to-peripheral processor and central processor-to-central processor).

9.23 The fault-analysis and fault-report routines analyze detected faults and perform the following functions:

(a) Rerun the fault-detecting test or a different test.
(b) Report the fault to the attendant console where it is indicated by an alarm display.
(c) Stores the fault data in an error log.
(d) Initiates a system reconfiguration.

9.24 The system-reconfiguration-and-call-recovery routine performs the following functions:

(a) Reloads the peripheral processor after a peripheral processor fault.
(b) Transfers control to the standby CEC if a fault has been detected in the on-line CEC.
(c) Saves all established two-way and three-way calls.

9.25 The software residing in the peripheral processor is divided into the following subprograms:

(a) PEC-control subprogram.
(b) Hardware-interface subprogram.
(c) Central-processor-interface subprogram.

Peripheral Equipment Cabinet Control Subprogram

9.26 The PEC-control subprogram is activated by interrupts from the on-line central processor. This subprogram maintains real-time control and is divided into the following two schedulers:

(a) Foreground scheduler.
(b) Background scheduler.

9.27 The foreground scheduler consists of a dispatcher that periodically schedules the following routines:

(a) Fast-access trunk-scan routine.
(b) Line-scan routine.
(c) Console-scan routine.
(d) DTMF-receiver-scan routine.
(e) DP-digit-collection routine.
(f) DP-digit-outpulsing routine.
(g) Ringing-control routine.

9.28 The background scheduler consists of a dispatcher that schedules routines to do the following:

(a) Perform a slow trunk scan and memory test every 100 milliseconds.
(b) Process directives from the central processors.
(c) Process information obtained by the peripheral processor foreground scheduler.

Hardware Interface Subprogram

9.29 The hardware interface subprogram performs the following subprograms:

(a) Scans hardware (lines, trunks, attendant console, DTMF receivers) and reports any changes as events to the central processor.
(b) Performs hardware operations (such as answer supervision and ringing control) as directed by the central processor.

Central Processor Interface Subprogram

9.30 The central processor interface subprogram provides the connecting link between the peripheral processor and the on-line and standby central processor via a common memory in each central processor. This subprogram is responsible for the following operations:

(a) Reports events to each central processor.
(b) Unloads directives from each central processor and passes them on to the appropriate processing application programs.

Line Control Subprogram

9.31 The line-control subprogram monitors the lines for on-hook, off-hook, and flash conditions, and reports any activity to the call control subprogram. The line-control subprogram assigns the digit stores and DTMF receivers to the lines when needed.

9.32 Digit Stores. In addition to the call stores, there are digit store memory blocks that are dynamically allocated upon processing of a line seizure, and linked to the associated call store during digit accumulation and outpulsing. The digit store memory contains the control words required to accumulate, store, and outpulse up to 16 digits.

Station Features Subprogram

9.33 The station-features subprogram performs the station features in conjunction with the digit-analysis and call-control subprograms.

Trunk Control Subprogram

9.34 The trunk control subprogram processes incoming and outgoing trunk seizure, incoming and outgoing trunk signaling, and trunk release information in conjunction with the call control subprogram. The trunk control subprogram also allocates digit stores and DTMF receivers to trunks when needed.

System Features Subprogram

9.35 The system features subprogram performs the system features under the control of the digit analysis subprogram and reports to the call control subprogram. The following features are executed by this subprogram:

(a) Intercept.
(b) Class-of-service checks.
(c) Station hunting.
(d) AIOD.
(e) Automatic Call Distribution (ACD).

Centralized Attendant Service Branch Subprogram

9.36 The CAS branch subprogram monitors seizure, signaling and release of the Release Link Trunks (RLT's) in conjunction with the call control subprogram. The CAS branch subprogram also allocates digit stores and DTMF receivers for the RLT's.

9.37 The CAS branch subprogram also performs the following CAS related features:

(a) CAS attendant camp-on and recall.
(b) CAS attendant recall on no-answer.
(c) CAS attendant silent hold.
(d) RLT night answer control.
(e) RLT direct and group access.

Message Detail Recording Subprograms

9.38 Message Detail Recording (MDR) provides the means for recording call information pertaining to incoming and outgoing trunk calls. As the trunk calls proceed through the system (GTD-1000), MDR obtains various pieces of information about the calls. Message Detail Recording (MDR) analyzes this information and stores the results in a call record block. The call record block can then be sent to any combination of three outputting devices.

9.39 The first output option is a cartridge type. If this option is selected, the call is recorded on a four-track tape cartridge. The tape must be sent to a service bureau that will process the tape and return a listing of the calls.

9.40 The remaining two options are terminal types. With these options, call record blocks are obtained from a dual port serial device controller card consisting of two ports.

9.41 Port 0 provides an Input/Output (I/O) port for a local TTY. As each trunk call is terminated, it is immediately printed out, providing a hard copy of the call record.

9.42 Port 1 provides an I/O port that will interface with one of the remote polling units or with a mini computer.

Health-Care/Motel Subprogram

9.43 The H/M subprogram handles all operations of the following devices and features:

(a) Key Entry Display Unit (KEDU) operation.
(b) Printer operation.
(c) CLR trunk operation.
(d) Wake up processing.
(e) Message waiting processing.
(f) Health-Care/Motel (H/M) feature activation/cancellation via phone.
(g) Health-Care/Motel (H/M) features on attendant console.
(h) Health-Care/Motel (H/M) dynamic data back up and read-in.
(i) Message meter pegging.
(j) Health-Care/Motel (H/M) feature processing on calls.
(k) Calling number display until processing.

10. TYPE 200 DIGITAL SYSTEM TEST SET

Description

10.01 The Type 200 Digital System Test Set (Figure 12) provides a man/machine interface for performing maintenance, verifying system operation, and isolating hardware and software problems. It also provides a means for easily and rapidly loading the system data base. For a detailed description of the test set, refer to Section 278-100-102.

Figure 12. Type 200 Digital System Test Set.

10.02 The test set plugs into an operating system via the TPI without interference with normal operation of the system. Changes and initial programming can be performed via a teletypewriter or TERMINET® machine.

10.03 The test provides a LED display of the contents of a memory word and its address. It also provides the controls for activating the following:

(a) Address. The test set can, via a 16-key hexadecimal keyboard, address any number from 0 to 64,000 for the purpose of reading or writing from that address.

(b) Functions. Via multiposition selector switches, the test set is capable of initiating the following functions:

(1) Single read.
(2) Single write.
(3) Read with automatic address advance.
(4) Write with automatic address advance.
(5) Continuous read.
(6) Address compare and stop.
(7) Single-instruction execution.
(8) Activation of software utilities.

Data-Base Loading

10.04 A software commercial program generator is supplied by GTE AE that allows a customer-provided data base to be prepared on tape and loaded into the system. On initial activation, approximately 22,000 words of data-base activation are loaded into the system's RAM. While it is not desirable, it is possible to input a data base into the system manually via the Type 200 Digital System Test Set or the TTY.

10.05 When a small data-base change, such as adding or deleting lines or changing line classes, is required, either the test set or TTY inputs the required change into an active system. The new data base is then loaded on the cartridge magnetic tape without interruption of call processing.

11. SYSTEM OPERATION

11.01 The sequence of events initiated by a station activation (receiver off-hook) in making a station-to-station call and the progression of that request for connection through the various subsystems (Figure 13) of the system are described in the following paragraphs.

11.02 Requests for service other than station to station are handled by the system. The call is either controlled by the applicable program or subprogram or forwarded to the operator for assistance or action.

11.03 Requests for services that are optional are discussed in the following paragraphs.

Call for Service

11.04 Several components are required to complete a call for service. Among these are the channel memory, the information memory, the control memory, suitably processed tones from the digital tone card, the digital pad, the system software (that is being processed by a peripheral processor or the central processor), and the network interconnect memory.

11.05 Each PEC contains 4 files of equipment and has an associated 96-location channel memory. Thus, each file has 24 specific locations (time slots) in its respective channel memory.

11.06 The first channel memory is associated with the even-numbered locations of the information memory (0,2,4,6,8, etc); the second channel memory is associated with the odd-numbered locations of the information memory (1, 3, 5, 7, 9, etc).

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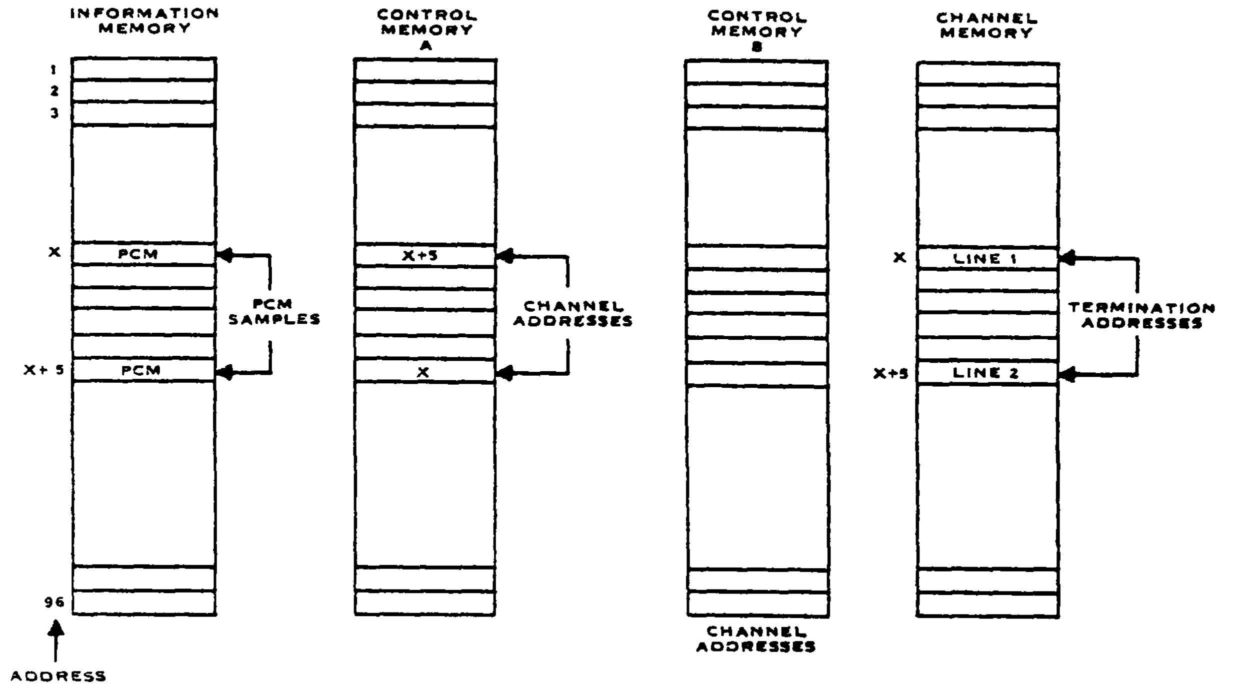

Figure 13. Network Memory Contents.

11.07 The data base is programmed to dedicate each piece of equipment in each PEC to specific equipment in other parts of the system. To enable a particular piece of equipment, the CPU writes the equipment address into the associated channel memory location.

11.08 The information memory contains 512 locations and is divided into 2 equal parts or networks. Each network is used to serve two PEC's. Of the 256 locations in each network, 192 locations are used for actual call processing time slots. The remaining 64 locations are used for digital tones. One information memory location is associated with each channel memory location and functions as storage for each PCM signal when the appropriate time slot comes up.

11.09 The network interconnect memory contains 512 locations and is divided into 2 equal parts. Each part contains the exact information contained in the corresponding part of the information memory. The network interconnect memory is used to interconnect calls between the two networks.

11.10 Digital tone values are continuously written into locations 192 to 255 of the information memory. When required by the CPU, a tone value is read from one of the 64 time slot locations. Separate signals or tone generators are therefore not required.

11.11 The control memory contains 2 memory sections (A and B), each with 384 locations. Section A is used to process regular calls; section B is used for three-way calling.

11.12 The function of the control memory is to store the information memory addresses of the stations desired. In a two-way connection, station A's PCM signal is stored in location 1 of the information memory, and station B's PCM signal is stored in location 2 of the information memory. The CPU reads the PCM signal from each of the two locations and writes the signal into the opposite location. This PCM signal is sent through the assigned channel to the station via the DTA converter. If both stations are not located in the same network, the CPU reads the PCM signal from the corresponding network interconnect memory location.

11.13 The digital pad used in the system is physically located on two separate cards: the pad control on the control-memory card, the pads on the information-memory card. Four pads of differing dB levels (Q2,3, and 5 dB) are available and are selected on a class-of-service type-of-call basis.

11.14 A comparator circuit seeks the PCM signal with the highest dB level in the information memory associated with the call, and this dB value is padded. When the information memory address of a voice sample is written into an information memory location, it is associated with a location in control memory. In the same memory location in the pad memory, the CPU writes the correct pad to be enabled. The digital value of the PCM signal in the

information memory is used as an address to access a port in the digital pad of the same PCM pad value. The PCM pad then sends a PCM signal of a value reduced by the amount of the pad. For example, if a PCM value is 30 dB and a 5-dB value was selected, the outgoing PCM value would be 25 dB.

11.15 The sequential steps of the processing of a call are as follows: When a call for service is initiated, the system software recognizes a change of state from on-hook to off-hook by comparing sequential look scans of the sense points. The system software identifies a calling station by knowing what sense points were activated and identifying the sense point to a hardware equipment number in a software table that lists this relationship. The equipment-identification number is also associated with the actual directory number.

11.16 The system software then selects an idle channel from the file containing the activated sense point. A time slot is assigned by writing in the equipment identity associated with the sense point in the channel memory. If a channel is not available, the search is repeated on a delay basis until one is found.

11.17 Since a one-to-one relationship exists between the information and control memories, establishing an equipment identity reserves a location in both memories.

11.18 Class of service is checked by the system software at this time to determine whether the calling party has DP or DTMF equipment. If it has DTMF equipment, an idle DTMF receiver is assigned to the calling party. DP receiving is direct from the associated sense points.

11.19 The system software then writes the information memory address of dial tone into the calling party's control memory location. When the calling party's time slot appears, nothing is written into the information memory during the first one-fourth cycle. During the second one-fourth cycle, the time switch again looks at the PCM signal stored in the information memory. At this time, it would be dial tone.

11.20 Dial tone is returned to the calling party through the DTA converters until the calling party dials the first digit, times out, or goes on-hook.

11.21 As soon as dialing begins, the system software clears the calling party's channel-memory location of dial tone by writing the address of quiet tone into that location.

11.22 In the case of DTMF receivers, when the time slot assigned to the DTMF receiver occurs, the DTMF receiver, via the ATD converter, receives the PCM signal of the digit stored in the calling party's information-memory location. This PCM signal is then sent as a digit for further analysis.

11.23 When dialing is completed, the system software checks to see if the called station is busy. If the called station is idle, the system software reserves a path to the called station by writing the called station's address into the channel memory. If a channel is not available, busy tone is sent to the calling station. If a channel is available, ringing current is sent to the called station.

11.24 When the called station goes off-hook, the system software receives the off-hook signal from the called station, writes the called station's information-memory address into the calling station's control-memory locations and then writes the address of the calling station into the called station's control-memory location.

11.25 The action by the system software removes ringback tone from the calling station and completes the connection. When conversation is completed and either station goes on-hook, all system resources associated with the call are cleared by the system.

Voice Information Transmission

11.26 Voice information (analog signals) are routed through the PCM sampler unit, which samples incoming analog signals at an 8-kHz rate. The resulting PAM sample is then encoded into an 8-bit binary code representing 128 distinct quantized amplitude signal levels by the ATD circuit.

11.27 The time-switch network clock generates timing pulses that scan the channel memory at 125-microsecond intervals and sequentially enable the lines identified by the addresses in the channel memory. The data from each line is then written into the information memory.

11.28 The channel and information memories working in conjunction with the control memory provide software control and are used by the CPU to transmit tone and voice information. Each word written into a specific location in the control memory relates to a corresponding word and location in the channel and information memories.

11.29 The following paragraphs describe a simplified sequence of events for the transmission of voice information between station 1 and station 2 (Figures 13 and 14).

11.30 The equipment number for the calling station (station 1) is written into the time slot X location in the channel memory, and the called-station sense-point-identity number (station 2) is written into the time slot location identified as X + 5.

11.31 The early counter sequentially reads the contents of the channel memory locations at the beginning of each time slot. When the CPU, under control of the call-processing subprogram, reads the contents of the time slot X location of channel memory, it reads the sense-point-identity number of the calling station (station 1).

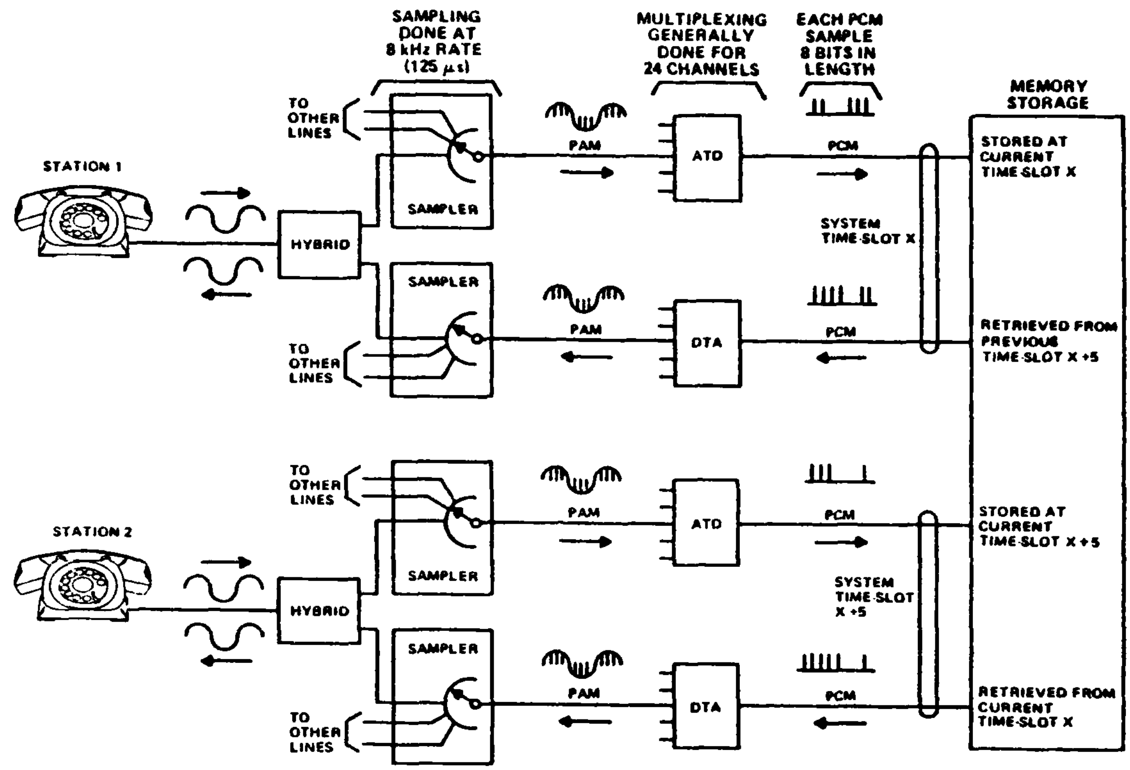

Figure 14. PCM Switching Fundamentals.

11.32 The channel enable circuit enables station 1 for voice transmission. The PAM sample is then routed to the ATD-converter circuit and converted into an eight-bit parallel sample. This sample is written into the time slot X location of the information memory.

11.33 Following the writing of the station 1 PCM sample into the information memory, the CPU, under control of the call-processing subprogram, reads the contents of the corresponding time slot X location from the control memory and obtains the address (X + 5) of station 2.

11.34 At memory address X + 5 of the information memory, the previous PCM sample from station 2 is read from the memory and routed to the DTA converter. Here, the eight-bit-parallel PCM input is converted to a PAM signal and transmitted to the line circuit. At a time determined by the channel-enable circuit, each line circuit samples and filters the PAM bus and transforms the PAM sample to an analog signal. The receive gate of station 1 is enabled at this time and station 1 receives voice transmission from station 2.

11.35 The call-processing program sequentially processes the data in subsequent time slots of the channel memory until it reaches the time slot X + 5 representing station 2. The sequence of steps used in processing the data in station 2 are repeated, causing station 2 to receive the voice intelligence transmitted by station 1.

11.36 When either station goes on-hook, the scan program marks the circuit. If the on-hook indication is detected on the next scan, the line is disconnected. The CPU then drops the time-switch-network connection and, with the associated call store, returns the disconnected station to an idle status. It marks the other station as waiting for disconnect. When the second station goes on-hook, it is returned to an idle status. If the second station remains off-hook, it goes into lockout.

APPENDIX B

EXCERPTS FROM GTE TELENET

PRIVATE DATA NETWORK PRODUCTS

FUNCTIONAL DESCRIPTION

PE-TP-002-02B

APRIL 1980

APPENDIX B

TABLE OF CONTENTS

4 TELENET PROCESSOR SYSTEMS

The following three major sections give detailed presentations of the Telenet Processor (TP) hardware architecture and components, the TP software architecture, and a summary of the TP product line.

4.1 TP HARDWARE ARCHITECTURE AND COMPONENTS

The basic architecture of a non-redundant TP is given in Figure 4-1. The principle components of a TP are the main memory bank, the arbitrator, the Central Processing Unit (CPU), the data and address busses, and the Line Processing Units (LPUs). Both LPU and CPU cards contain an MOS Technology 6502A microprocessor, all of which run asynchronously with respect to one another. TP software (see Section 4.2) was developed to take full advantage of this quasi-parallel operation by functionally partitioning the processing load among the different microprocessors. Basically, the CPU is responsible for processing the TP Operating System (TPOS) and for processing the X.25 packet level protocol for all virtual circuits handled by the TP. The LPUs provide access to the communication lines and process the X.25 link access procedures or Packet Assembly/Disassembly (PAD) software.

The microprocessor used has an address field of 16 bits. This gives access of up to 64K bytes of memory which is mapped into a local or main memory area. Each microprocessor card is equipped with either 8K or 16K bytes of dedicated local memory which is used to store the frequently accessed portion of the processor's program. This local memory can be accessed without contention from other processors. Main memory ranges in size from 64K bytes to 256K bytes and is primarily used to hold copies of the LPU local memory code, main memory code, system tables (e.g., routing tables), and for buffering of traffic (user data). An arbitrator unit controls accesses and contention to main memory by the CPU and LPUs. The arbitrator will allow the CPU(s) to use up to 50% of the memory bandwidth with the remaining 50% sequentially shared among the LPUs. All accesses are made on a demand basis; that is, if on a particular turn a processor does not require a memory fetch (or write), it is bypassed and its memory cycle is given to the next processor in the polling sequence. This polling activity normally occurs as a parallel function to memory accesses and does not add any significant delay. The memory and arbitrator units are connected to the CPU and LPUs by an address and data bus, both of which are parity protected to ensure data integrity.

BASIC TP ARCHITECTURE

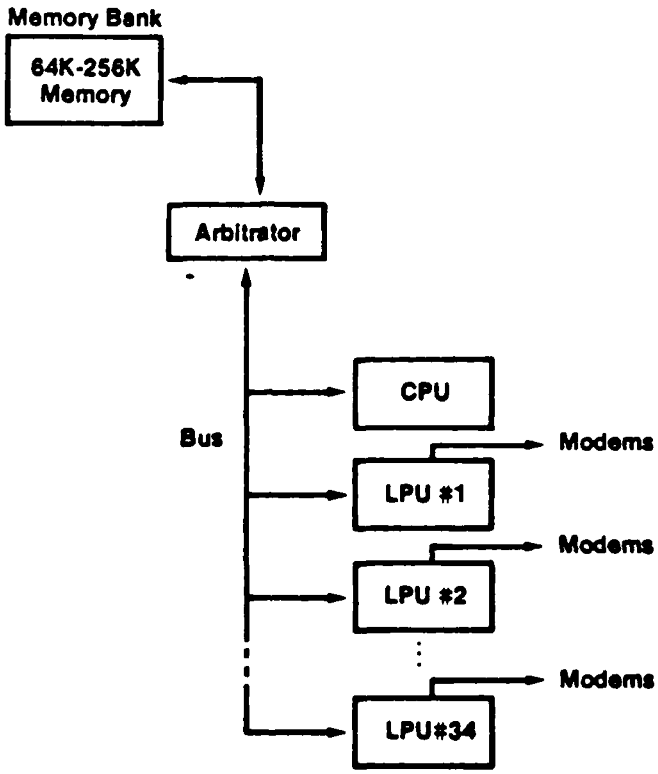

Figure 4-1

Certain Telenet Processor models (see Section 4.3 for details) can be equipped with redundancy to overcome any single unit failure without a permanent loss of capability. The architecture of a redundant TP is given in Figure 4-2. TPs can be configured with two types of redundancy; common logic redundancy and 'One for N' LPU redundancy, and are equipped only with a 128 or 256K byte memory.

REDUNDANT TP ARCHITECTURE

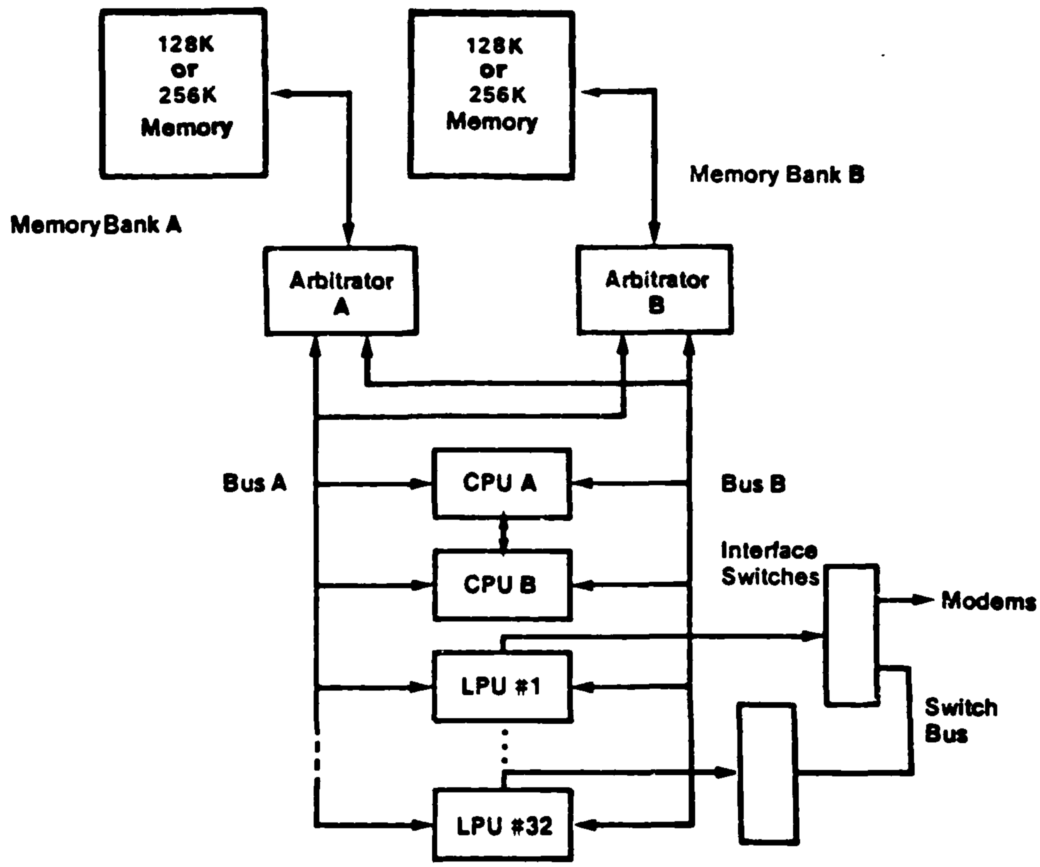

Figure 4-2

In a system with common logic redundancy all centrally used components are duplicated. The CPU, memory bank, arbitrator, power supplies, and busses each have a back-up unit which is automatically switched into service when the active unit fails. The memory, busses, and arbitrator units are treated as a single component since the failure of any device causes the switchover of all three back-up units.

Switchover to a back-up component may cause a temporary loss in service and clearing of any calls originating and terminating on the TP. Switchover times range from instantaneous for a power supply, to several minutes for a memory bank switchover. All back-up components are constantly monitored by TP software for errors to avoid the situation of switching to an already failed unit.

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SEP 80 M ROSS, K GARRIGUS, J GOTTSCHALCK DCA100-79-C-0024
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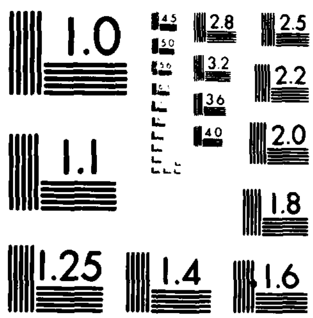

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'One for N' redundancy is designed to provide a back-up capability for the line processing units. For a group of N LPUs of a particular physical type, one LPU of the same type is the single common back-up for each LPU in the group. This is accomplished by installing LPUs with logic switches that can route the LPU communication lines to a back-up LPU. When an LPU has failed, the system CPU will automatically activate the logic switches, which transfer the lines to the back-up. The back-up LPU's local memory is then loaded from main memory, where a single copy of all LPU local code is kept, and the unit initialized. All calls originating or terminating on the failed LPU will be lost but may be re-established by the user after the back-up card is operational. This takes approximately 30 seconds. As with the common logic back-ups, back-up LPUs are monitored for failures while in stand-by status. Once the back-up card is in use, any subsequent LPU failure within the group will cause loss of service. It is possible to have one back-up card for each active LPU card, affording total system availability. However, the calculated high reliability of LPUs makes simultaneous failures a very unlikely event and only a single back-up LPU, per LPU model, is recommended.

In the following sections the individual TP components are described in further detail.

4.1.1 Memory and Arbitrator Units

TP main memory is available in 32K byte, 128K byte, or 256K byte cards to a maximum of 128K or 256K bytes (per bank) depending upon the TP model. Data have full parity protection; additionally, the 128K byte and 256K byte memory modules are equipped with single bit error correction capability based on the Hamming code technique. If uncorrectable parity errors occur, the event is logged at the Network Control Center and the back-up memory, if available, is switched into operation. Note: The TP will first attempt to reload its memory. If an error in the memory is detected during this loading operation then an attempt will be made to switch and reload the back-up memory bank. If no error is found during the reload operation the TP will continue to use the primary memory bank.

Main memory is divided into the following six areas:

- a communications area which is used to exchange commands between the multiple microprocessors;
- a local code area where one unique copy of the LPU's local code is kept. This allows an LPU to load its local code area without help from the Network Control Center. The local code area is used wherever an LPU is restarted or a back-up LPU is switched into service;
- a main memory code area which contains all the code that is directly executed by the CPU and LPUs (main memory code is write protected since it is shared);

- a table area which includes routing tables, terminal ID tables, ITI tables, etc.;

- a buffer space area used for the temporary buffering of user traffic, dynamic tables, and accounting information. The buffer area is broken down into full buffers of 144 bytes of which 128 bytes can be used to transfer data. A full 144 byte buffer can be further subdivided into quarter buffers for terminal traffic buffering. Buffer management and garbage collection are automatic features of the TP Operating System (TPOS); and,

- a 4K byte Read Only Memory (ROM) device which contains a single channel implementation of the X.25 protocol and sufficient information to remotely load the system both when the TP is initially powered up, and when the system is recovering from a major failure.

Although main memory has an access cycle time of 450 nanoseconds, the effective access rate by any given microprocessor can be approximated at 2 to 3 microseconds due to memory and bus contention.

The arbitrator unit resides on a single card which operates in conjunction with main memory. The arbitrator has the responsibility of controlling all accesses to main memory from either bus. As described earlier, up to 50% of main memory bandwidth is assigned on demand to the CPUs. Should the CPUs not utilize their total bandwidth, the unused bandwidth is dynamically re-allocated to the LPUs. In addition to its control function, the arbitrator is responsible for system services such as memory error control features and an Automatic Memory Tester (AMT). The AMT operates in background mode and continuously cycles through main memory checking for parity errors, thus providing an early warning of memory failures. The arbitrator contains 16 strap bytes which are used by the 4K byte ROM to identify the TP in the call establishment procedures to the NCC during system initialization. The arbitrator is also responsible for handling intraprocessor interrupts, and contains a FIFO stack for queuing multiple quasi-simultaneous CPU interrupts arriving from the LPUs. The stack mechanism ensures that the interrupts are processed in their order of arrival and that none are lost.

The active arbitrator, memory units, and busses are logically coupled and a failure in any of these devices causes the system to switch to all three back-up units.

4.1.2 Data Address/Control Busses

The arbitrator and memory units are connected to the CPU and LPUs through two busses; an address bus and a data bus. The address bus is 18 bits wide with two parity bits, while the data bus is eight bits wide plus one parity bit.

The parity of both busses is checked after each bus transfer, and if an error has occurred a retry is automatically initiated. If, after many retries, the error cannot be corrected and a valid transfer made, the system will produce a diagnostic message through the TP Reporting Facility (TPRF) and will switch to the backup bus, memory, and arbitrator set.

4.1.3 Central Processing Unit

The central processing unit (CPU) resides on a single card which contains an MOS Technology 6502A microprocessor, local memory, clocks and timers, and a local TTY interface.

The 6502A processor, also used in the LPUs, is a general purpose 8-bit microprocessor with an efficient instruction set which includes indexing, indirect, relative, absolute and immediate addressing modes. The 6502A has a microcycle time of approximately 500 nanoseconds when accessing local memory. An instruction may take between two and seven microcycles to complete depending upon its type. When code is executed out of main memory, microcycle times of two to three microseconds are typical due to other processor contention.

The CPU card is equipped with 16K bytes of local memory and 1K byte of ROM. The remaining 47K bytes of its address space is dynamically mapped to main memory through the use of a special mapping technique. Main memory may contain up to 128 pages, each containing 2K bytes of storage. The processor can simultaneously access up to 24 pages by setting special Page Map Registers (PMRs) located in the local memory area. By dynamically altering the values of the PMRs, the processor can access all of main memory.

The CPU card is equipped with a high speed inter-CPU interface and a local TTY interface which is used for diagnostic purposes. The CPU is also equipped with a watch dog timer designed to automatically start a re-initialization of the card if a software or hardware failure occurs causing the timer to expire. Normally this timer is reset at least every 140 milliseconds.

4.1.4 Line Processing Units

Access to communication lines is provided by Line Processing Units (LPUs). Each LPU has its own 6502A microprocessor, 8K or 16K bytes of local memory, and supports either one, four, or eight lines, depending on the speed of the line(s) and the protocol employed. The LPUs use the same paging mechanism as the CPU for accessing main memory, i.e., 32 PMRs. LPUs utilize main memory primarily for data transfers and buffering.

Telenet currently offers the following six LPU models:

Feature	LPU Type	Lines	Interface Type
111	Asynchronous	4	RS-232
112	Asynchronous	8	RS-232
122	Asynchronous	8	Current Loop
211	Synchronous (BSC)	4	RS-232
231	Synchronous (BSC)	1	V.35
212	Synchronous (HDLC)	8	RS-232

Telenet can also provide 4-line interface adaptors to convert RS-232 to V.35 or vice versa.

The throughput capabilities of any LPU are highly dependent upon the LPU type and the protocol employed as described below.

4.1.4.1 Asynchronous LPUs

The asynchronous LPUs will normally be loaded with the Interactive Terminal Interface PAD. If loaded with special customer software, the following figures may not be applicable.

The total cumulative throughput of the low speed asynchronous LPUs (LSLPUs) is 9600Kbps, that is, the aggregate throughput of the lines utilized cannot exceed 9.6Kbps. For example, an 8-line LPU can have all eight lines operating simultaneously at an average data rate of 1200bps to reach this 9.6Kbps limit. Line speeds can be mixed on a card as long as the total throughput is equal to or less than 9.6Kbps. The maximum speed of any one line is 2.4Kbps.

4.1.4.2 Synchronous LPUs

The throughput limitations of medium speed synchronous LPUs (MSLPUs) varies by the LPU model and the protocol.

The four-line LPU can be used to support either HASP or 2780/3780 traffic. When used with these protocols, the four-line card can support a total cumulative throughput of 38.4Kbps with an individual line speed limit of 9.6Kbps.

When used to support the X.25 protocol, the one, four, and eight-line cards can support the following throughputs:

Feature	Lines	Total Throughput	Single Line Maximum
211	4	38.4Kbps*	9.6Kbps*
231	1	56Kbps	56Kbps
212	8	224Kbps	56Kbps

* The total and individual line throughput on this card is reduced to 19.2Kbps when this LPU is also used as a system master (see Section 4.2 below).

The four and one-line cards support only BSC X.25 framing. The eight-line card supports only HDLC X.25 framing.

4.1.5 Environmental Requirements

The environmental requirements for a TP are as follows:

Power Requirements

120VAC, 60 Hz, 1580 Watts. Requires one separately fused 20A, 120VAC circuit with a three-prong duplex outlet (NEMA #5-20R), for each TP 4000 chassis in the cabinet. EXAMPLE: Dual chassis redundant TP 4000 cabinet (two chassis) requires two separately fused circuits.

Space Requirements

The TP 4000 C, H, and S models require one or two cabinets 72" high, 24" wide, and 30" deep. The TP 4000L and TP 2200 require a single cabinet 48" high, 24" wide, and 30" deep.

Heat Dissipation

5400 BTU. Requires sufficient environmental control to keep ambient air temperature between 32-104 degrees F.

4.2 SOFTWARE ARCHITECTURE

The software architecture of the TP is based on the inherent layering of the X.25 protocol. TPs use a 'system master' design where one card acts as the central controller of the system. The master is responsible for executing the major part of the TP Operating System (TPOS) and the X.25 packet level procedures. The master (either a CPU or LPU) performs all virtual circuit switching. The rest of the processor cards (LPUs) are called 'slave' cards and either primarily process the X.25 frame level procedures or higher level PAD software. In either case packets are built in main memory and passed to the master for processing. Thus, different X.25 link access procedures (LAP and LAPB framing, for example) are implemented on different LPUs and can still use the common master X.25 packet procedures. In general, slave cards provide protocol translation from some native protocol to X.25 packet level and vice versa. Slave cards also execute a minor portion of TPOS.

In the smaller TP concentrator systems (TP 2200 and TP 4000L) the master is a four-line synchronous LPU card, of which only one line is utilized.

This LPU master not only performs the previously mentioned functions but also provides the network trunk line interface. The basic software architecture of this type of system is given in Figure 4-3. Use of an LPU master presents three major system limitations. First, LPU master systems cannot be configured with redundancy, (common logic or 'One for N') since special capabilities found only on a CPU are necessary. The second and third limitations deal with throughput; in the LPU master system, total line throughput is limited to 19.2Kbps. The last limitation is that on an LPU master system the packet throughput for the TP is limited to a maximum of 25 packets per second.

In the larger TP systems the master role is filled by a CPU card (see Figure 4-4). In these systems the X.25 trunk line interface is provided by a dedicated LPU card. The three limitations found in an LPU master system (redundancy, line throughput and packet throughput) are now lessened.

LPU MASTER SYSTEM

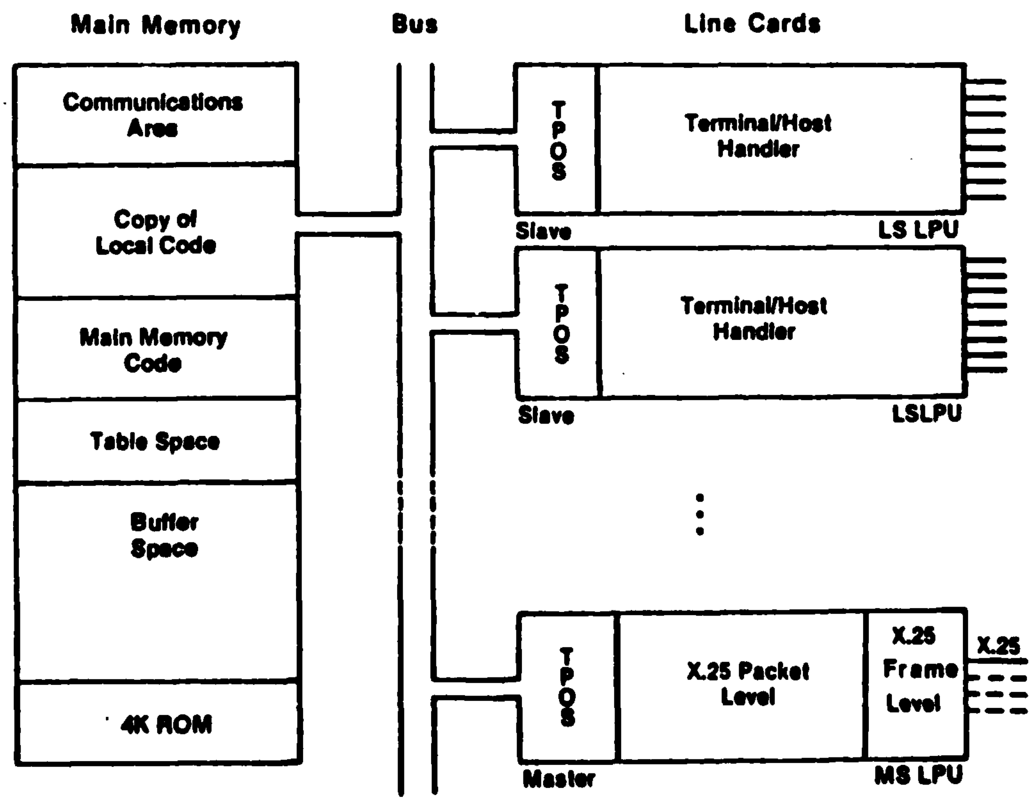


Figure 4-3

CPU MASTER SYSTEM

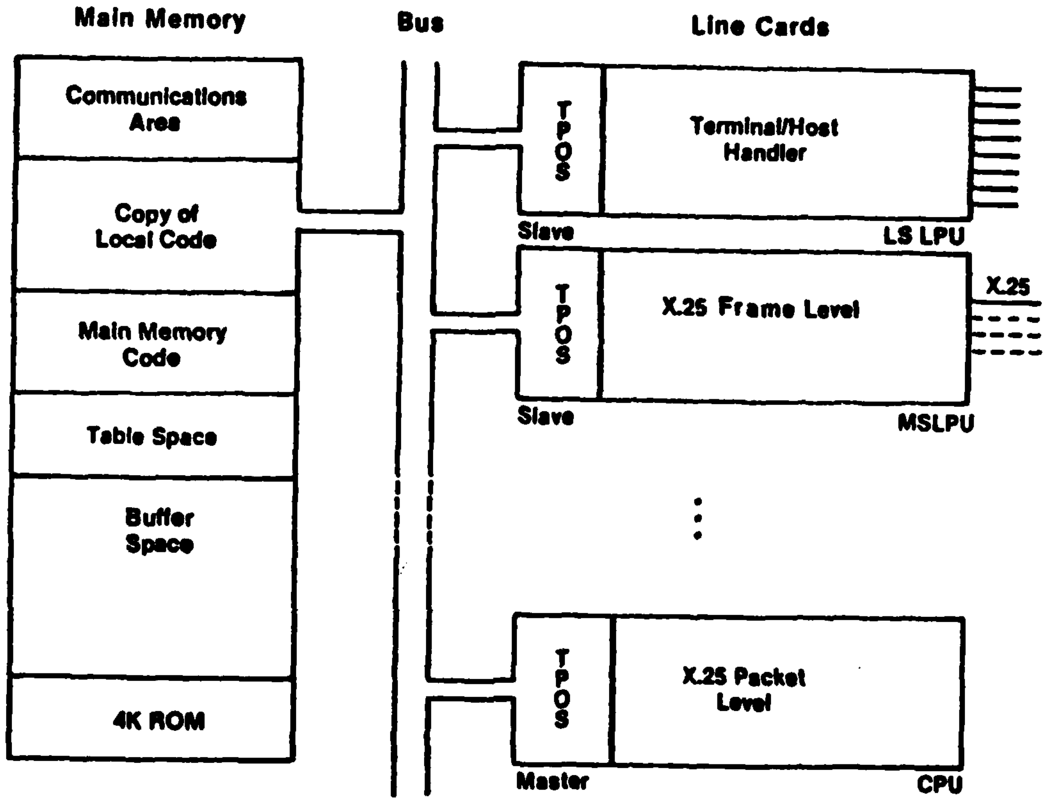


Figure 4-4

4.2.1 Telenet Processor Operating System

The Telenet Processor Operating System (TPOS) performs four major functions:

- Inter-card communication
- Buffer management
- Redundancy control
- Event reporting and diagnostic controls

The inter-card communication function is responsible for the exchange of commands between the master and slave cards. Typical operating system commands are initialization commands, buffer request commands, and buffer transfer commands. There is also a "hello" command which is sent periodically from all slaves to the master to verify that the slave is still operating. There are other commands that perform X.25 functions; for example, sending data, interrupts, and resets. Other commands allow the establishment and clearing of virtual circuits.

The second major function of TPOS is management of the buffer space or pool.

The third major function of TPOS is to provide the software support for the redundancy features.

The last major function of TPOS is support of the network management functions within the TP; specifically, the reporting of events to the NCC TP Reporting Facility and support for the Telenet Diagnostic Tool. For further detail see Section 5 on Network Management.

4.3 TP PRODUCT LINE SUMMARY

The TP product line consists of five models: the TP 2200, TP 4000L, TP 4000C, TP 4000H, and TP 4000S. Of these five models, four are concentrators and one (the TP 4000S) is a packet switch.

Concentrators provide the ability to concentrate traffic to and from many sources into one or more trunk lines. They do not switch local traffic or dynamically route traffic transiting the node. Hence, all traffic originating in a concentrator, destined for itself or another, must be routed through a switch.

The TP 4000S model provides all of the capabilities of the concentrator models, plus the ability to dynamically establish virtual circuits between itself and any other switch or concentrator to which it is linked.

The basic functional specifications for each of the TP models are included in this section, following Tables 4-1 through 4-3.

Tables 4-1 through 4-3 list the capabilities and memory requirements of each TP model.

FEATURES	TP2200	TP4000L	TP4000C	TP4000H	TP4000S
Trunk Lines					
Maximum Line Speed	19.2Kbps	56Kbps	56Kbps	56Kbps	56Kbps
DTE Interfaces					
Asynchronous host (ITI)	<=64	<=64			
Asynchronous terminal (ITI)	-		<=256	<=256	<=256
Block mode host/terminal	-	-			
(BMTI)	-	-			
X.25	-	-	-	yes	yes
Optional Redundancy Features					
Network Access Line	yes	yes	yes	yes	yes
One-for-N LPU	-	-	yes	yes	yes
Common Logic	-	-	yes	yes	yes
Other Features					
Event Reporting	yes	yes	yes	yes	yes
Virtual Call Reconnect	-	-	-	yes	yes
Full Call Accounting	-	-	-	-	yes
Interface cascaded TP concentrators	-	-	-	yes	yes
Local and transit switching	-	-	-	-	yes
Processing capability (packets per second)	25	25	125	225	225
Field upgradeable from models	-	-	-	C	C & H
LPU Master	yes	yes	-	-	-
CPU Master	-	-	yes	yes	yes

Table 4-1
TP Features

FEATURES	TP2200	TP4000L	TP4000C	TP4000H	TP4000S
<u>Memory Size</u>					
Minimum Memory Size (bytes)	64K	128K	128K	128K	128K
Maximum Memory Size (bytes)	128K	128K	256K	256K	256K
<u>Main Memory Utilization</u>					
Minimum Memory (bytes)	64K	64K	128K	128K	128K
Master System Size (bytes)	32K	34K	38K	74K	74K
Available Memory for Protocol Software and/or Buffers	32K	94K	90K	54K	54K

Table 4-1
TP Features

PROTOCOL	AVAILABLE MEMORY CONSUMED
Asynchronous Host	10K bytes
Asynchronous Terminal	36K bytes
2780/3780	26K bytes
MLI (HASP)	24K bytes
X.25 Link Procedure	16K bytes*

Table 4-2
Protocol Size Considerations

*Only required in the TP 4000H and S if both byte-oriented and bit-oriented procedure is required. Otherwise the memory required for one procedure is included in the master system size above.

PROTOCOL	TP 2200, TP 4000L & C	TP 4000H & S
Asynchronous	221 per port	381 per port
Synchronous	1418 per port**	1578 per port**
X.25 host	N/A	284 per active VC for interactive traffic 1076 per active VC for batch traffic

Table 4-3
Buffer Size Considerations

**Assuming BSC block size of 256 bytes.

4.3.1 TP 2200

The TP 2200 is one of two models in the TP product line that is configured without a CPU module. In this model, the medium speed line processing unit servicing the trunk line also functions as the system master.

The TP 2200 is a host interface processor designed to concentrate asynchronous interactive host traffic into one or more X.25 trunk lines connected to a TP switch (see Figure 4-5).

HOST INTERFACE PROCESSOR

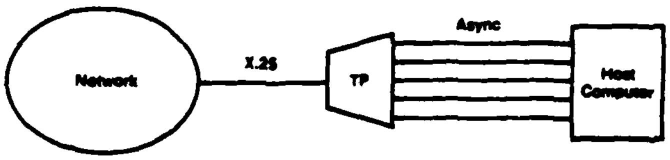


Figure 4-5

Note that normally all concentrators are directly connected to a TP 4000S switch; however, they may be optionally connected to a TP 4000H in a cascaded configuration (see Section 4.3.3).

The TP 2200 can be configured with up to eight low speed LPUs which will provide support for up to 54 host ports. The line speed of each X.25 trunk is limited to 19.2Kbps since a master LPU is used.

While no common logic or 'One for N' redundancy is available, there is the capability for network access line redundancy; that is, multiple X.25 lines from the TP to the network can be supported. This prevents a single modem failure or line outage from disabling the TP's functionality. Some virtual circuits will be cleared, however, since the virtual circuit load is balanced across all available links. It has complete event reporting and network management capabilities. The TP 2200 is equipped with two 32K byte memory boards and is field upgradeable to a maximum of 128K bytes in 32K byte increments. The TP 2200 is capable of processing up to 25 packets per second.

4.3.2 TP 4000L

The TP 4000L is the second model configured without a CPU module. The TP 4000L has all of the capabilities of the TP 2200 plus the ability to concentrate terminal asynchronous traffic (see Figure 4-6).

HOST/TERMINAL INTERFACE PROCESSOR

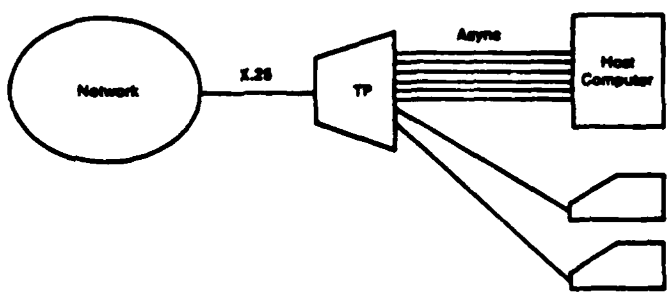


Figure 4-6

The TP 4000L can be configured to interface up to 64 asynchronous terminal or host devices with a maximum of eight LSLPUs. The TP 4000L is equipped with a memory of 128K bytes, which is the maximum allowable for this model. The TP 4000L is capable of processing up to 25 packets per second.

4.3.3 TP 4000C

The TP 4000C contains all of the functionality of the TP 4000L plus a number of additional capabilities. The TP 4000C can be used to concentrate synchronous BMTI host and terminal traffic in addition to asynchronous traffic (see Figure 4-7).

HOST/TERMINAL INTERFACE PROCESSOR

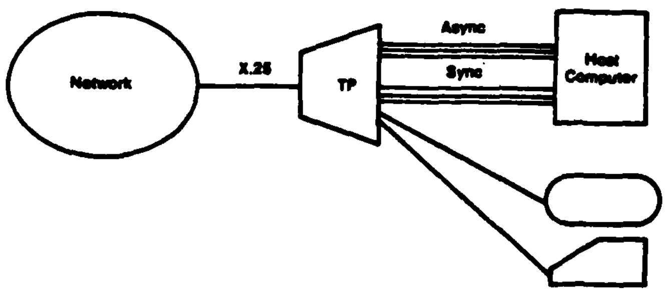


Figure 4-7

The TP 4000C is equipped with a CPU card which serves as the system master and is available in a redundant configuration of both common logic and 'One for N'. A maximum of 255 devices can be interfaced (see Section 4.3.6).

It is equipped with a single 128K byte memory card but can be ordered as a TP 4000CF with one single 256K byte card. Multiple network trunk lines can be configured using 56K byte circuits.

The TP 4000C can support up to 125 packets per second of traffic (see Section 4.3.6).

The TP 4000C can be field upgraded to a TP 4000H or TP 4000S.

4.3.4 TP 4000H

The TP 4000H is the most powerful concentrator in the TP product line. It can concentrate traffic from asynchronous ITI hosts and terminals, synchronous BMTI hosts and terminals, and to X.25 terminals. It also has the advanced capability of interfacing cascaded TP concentrators into a TP switch (see Figure 4-8).

CASCADED TP NETWORK

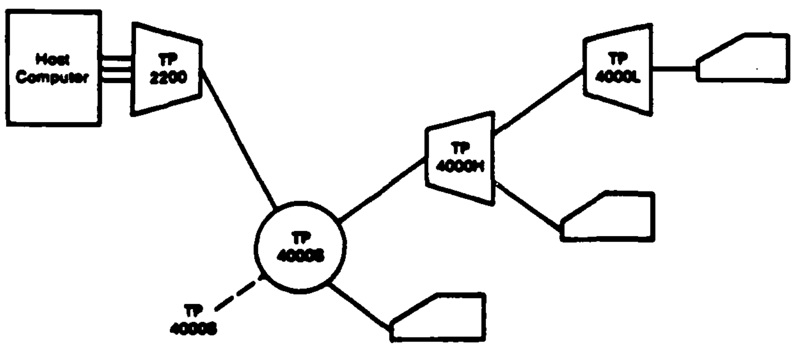


Figure 4-8

The TP 4000H has the same memory and LPU configuration rules as the TP 4000C (see Section 4.3.6 below).

The TP 4000H can support up to 225 packets per second of traffic. It can be upgraded to a TP 4000S.

4.3.5 TP 4000S

The TP 4000S is the one switch in the TP product line. It possesses all of the functionality of the concentrator models plus its switch capability. The TP 4000S can be used for local or transit switching as shown in Figure 4-9 below.

TP 4000S SWITCH

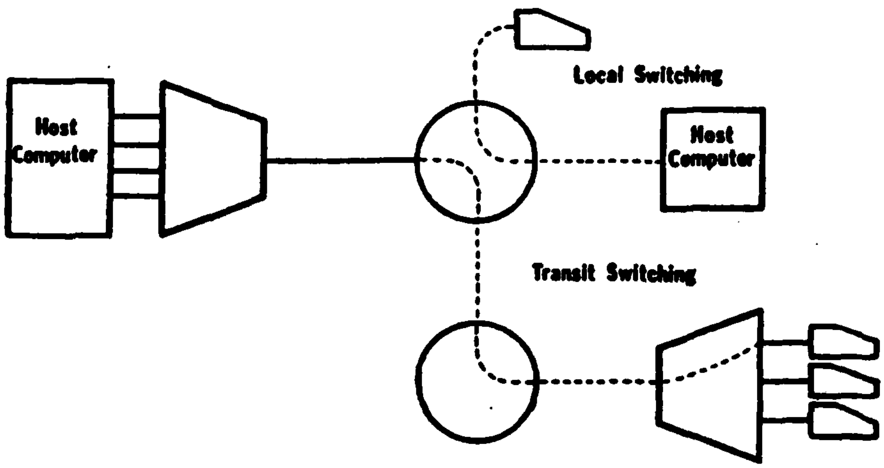


Figure 4-9

The TP 4000S performs the full call accounting procedures as described in Section 5 on Network Management. The TP 4000S is also responsible for initiating the reconnection of interrupted virtual circuits.

The TP 4000S has the same memory and LPU configuration rules as the TP 4000C and TP 4000H (see Section 4.3.6 below).

The TP 4000S is equipped with a single 128K byte memory card or can be ordered as the TP 4000SF with one 256K byte memory board.

The TP 4000S can support up to 225 packets per second of traffic.

A TP 4000S can be configured by upgrading a TP 4000C or TP 4000H.

4.3.6 Configuration Rules

A TP is constructed using one or more chassis, each of which can hold up to nine LPUs, four memory card(s), an arbitrator, and a CPU card.

The TP 2200 and TP 4000L are housed in a short (48") cabinet containing one chassis. They may be ordered with or field upgraded to a maximum of 9 LPUs (one master, eight slaves). These systems are not field upgradeable to larger systems.

The TP 4000 C, H, and S are housed in a full sized (72") cabinet containing two chassis and are equipped with a single 128K byte memory card.

The TP 4000 C, H, and S may be ordered with or field upgraded to include:

- an additional 128K bytes of memory (Feature 403) and up to 34 LPUs;
- redundant configuration with 128K bytes of memory (Feature 308) and up to 32 LPUs; or,
- redundancy (Feature 308) and memory increase (2 items of Feature 403) and up to 30 LPUs.

The TP 4000 F or "Full" models are equipped with one 256K byte memory board. The TP 4000 CF, HF, and SF may be ordered with or field upgraded to 34 LPUs. With the redundancy option (Feature 309) they may only include 32 LPUs.

To determine the actual memory required for a given TP application, refer to Section 7 on Memory Sizing.

4.3.7 Performance

In a packet switched network, the performance of a TP is a function of the utilization of the available processing power. Available processing power is defined as the total processing power of the CPU minus the amount needed to run the TP Operating System. The amount of processing time available for use in a TP is 750 milliseconds out of each second, with the remaining 250 milliseconds used by the operating system. As utilization of the available processing power

reaches a maximum, buffering requirements increase drastically and system performance decreases.

A utilization rate of 75-80% has been found to be the maximum limit for effective operations.

To calculate the TP processor utilization two factors must be known:

1) The types of traffic the TP will be handling. Traffic types are categorized according to the service required by the packet. Packets that just transit a TP require only switching from one inter-TP trunk line to another and are therefore considered trunk-to-trunk traffic. Packets originating or terminating at a TP require additional services such as padding and accounting along with switching and are considered one of the following traffic types:

   - trunk - X.25 host
   - trunk - port
   - X.25 host - X.25 host
   - X.25 host-port
   - port-port

2) The processing time needed to handle each packet. This time is a function of the type of traffic and is the same for all packets of a particular type, regardless of their length. Processing time increases as the number of services required by the packet increases. As an example, consider a TP 2200 host concentrator. The only type of traffic handled by the TP is between asynchronous host ports and trunk lines. The time required to service a packet of this type is 22.5 milliseconds. To maintain a processor utilization of 75%, the maximum number of packets per second the TP can handle will be 25 (750 ms. x .75/22.5 ms. per packet). At this level the system performance will be high with small delays. If the number of packets handled by the TP exceeds 25 per second, delays will increase and system performance will suffer.

Determining the processor utilization for any TP in a network requires a summation of the number of packets of each type multiplied by the service time required for each type of traffic. This will determine the amount of CPU time required to handle the traffic load. Dividing this total by 750 milliseconds of available processing time per second will yield the processor utilization rate for that particular TP. The time required for processing the different packet types is given in Table 4-4 below:

Function	ms/packet	max,pkts/sec	Kp/hr
Trunk-Trunk	1.5	400	1440
Trunk-X.25 host	2.5	240	864
Trunk-port (sync.or async.)	4.5	133	479
Trunk-port (TP2200 or TP4000L)	22.5	25	96
X.25host-X.25host	3.5	171	616
X.25host-port (sync.or async.)	5.5	109	392
Port-Port (sync.or async.)	7.5	80	288

Table 4-4
Typical Peak Loads

If processor utilization rate is above 80% steps must be taken to decrease the traffic load on the TP. Several options are available to accomplish this including the installation of additional TPs to share the traffic load, the installation of additional trunk lines to route traffic around the over-utilized TP, and changing routing tables to redistribute traffic throughout the network.

4.3.8 Standard TP Configurations

Each TP is provided in a standard (base) configuration, to which the user adds modules, such as LPUs and memory, required to support a unique traffic environment. Base configurations are:

TP 2200

Quantity	Feature	
1	1902	Chassis
1	1904	48" Cabinet
2	1402	32K byte memory
1	1401	Arbitrator
1	1404	Power Supply
1	1405	Blower
1	1906	EIA Connector Block

TP 4000L

Quantity	Feature	
1	1902	Chassis
1	1905	48" Cabinet
1	1407	128K byte memory
1	1401	Arbitrator
1	1404	Power Supply
1	1405	Blower
1	1906	EIA Connector Block

TP 4000 C, H & S

Quantity	Feature	
1	1902	Chassis
1	1905	72" Cabinet
1	1407	128K byte memory
1	1403	CPU
1	1401	Arbitrator
1	1404	Power Supply
1	1405	Blower
2	1906	EIA Connector Block

TP 4000 CF, HF & SF

Quantity	Feature	
1	1902	Chassis
1	1905	72" Cabinet
1	1408	256K byte memory
1	1403	CPU
1	1401	Arbitrator
1	1404	Power Supply
1	1405	Blower
2	1906	EIA Connector Block

REDUNDANCY OPTION 308 (128K)

Quantity	Feature	
1	1902	Chassis
1	1407	128K byte memory
1	1403	CPU
1	1401	Arbitrator
1	1404	Power Supply

REDUNDANCY OPTION 309 (256K)

Quantity	Feature	
1	1902	Chassis
1	1408	256K byte memory
1	1403	CPU
1	1401	Arbitrator
1	1404	Power Supply

5 NETWORK MANAGEMENT

Network management involves collecting accounting information, monitoring and diagnosing system malfunctions, and receiving event reports and alarm messages. There are three options available to a Telenet customer for the management of a private network.

A private customer may elect to utilize the full services of the Telenet Network Control Center (NCC). In this case, the private network is connected to the Telenet NCC by an X.25 internodal link, and all network management functions are handled by Telenet, with accounting information being supplied to the customer on a monthly basis.

The second option for a private customer is the Customer Network Control Console (CNCC). This allows the customer himself to monitor in real time the status of his network and to initiate repairs. Accounting information is still collected by the Telenet NCC and delivered to the customer.

Finally, the customer may be totally independent of the Telenet Network by choosing to have his own NCC, which would give him full network management capabilities.

5.1 NETWORK CONTROL CENTER

Management of a network is provided by the Network Control Center (NCC). The Telenet Network Control Center performs three basic functions that are critical to the reliable operation of a network:

- It is the command and control center to which failures in a network are automatically reported and the focal point for isolating these malfunctions and initiating the necessary remedial actions.
- It is the data collection vehicle for all accounting information for a network.
- It is the source for building and loading tables that describe the configuration of each Telenet Processor (TP) in a network. It is also the source for loading new software releases throughout the network.

The NCC consists of one or more minicomputers connected to the network via X.25 ports, i.e., it communicates with all TPs via X.25 virtual calls. For example, to down-line load software to a TP, the TP will transmit a CALL REQUEST packet to the NCC. The NCC determines which software systems are needed for both master and slave cards and sends them, along with the necessary configuration tables, to the TP. The TP receives and loads these programs, and control is turned over to the master card, which initializes the system.

The TP is fully operational when the local memory on each slave LPU is loaded from the TP's main memory.

As described previously, a network may be completely independent of the Telenet Network, if it is configured with an NCC. Alternatively, a network may be configured with at least one X.25 internodal connection to the Telenet Network. A Customer Network Control Console (CNCC) is then used to provide a customer with various levels of network management functions. The CNCC is an asynchronous terminal configured to time-share the Telenet NCC. In this manner, event reports and statistical information can be routed to the console by establishing a virtual call from the NCC to the CNCC. The console has further capabilities which allow an operator to query, test, and debug a private network.

5.1.1 The NCC Computer

The NCC computer, called the Network Control Processor (NCP), is a PRIME minicomputer running under the PRIMOS IV Operating System with several Telenet designed enhancements. The NCP is connected to the network via a BSC-framed X.25 link. It communicates with various elements in the network by using switched virtual calls.

The NCP is equipped with disc storage, magnetic tape, and communications ports for operator video and printing terminals.

The primary functions of the NCP are to accept status reports from each TP and determine the actual state of trunk lines and nodes. If network outage or failure is detected, the NCP immediately alerts the operations staff via console messages.

The NCP stores software and configuration tables for all TPs in the network. Upon request from a TP, this information is loaded via the network from the NCP into the TP's main memory. Other functions of the NCP include:

- the on-line debugging of specific network components;
- the gathering of accounting information; and,
- the statistical analysis of data that has been gathered.

The Telenet Network operates with multiple NCPs, so that in the event of failure of a single NCP, the NCC functions can continue to be performed. Note that the network can operate independently of the NCP (virtual calls in progress will not be interrupted, and new virtual calls can be initiated); that is, a malfunction in the NCP will not affect normal operation in the network, except for the following possible problems:

- accounting data and status information may be lost;
- debugging facilities may not operate; or,
- attempts by a TP to get software or configuration tables may fail.

5.1.2 NCP Support of TP Software

The NCP communicates with TPs via X.25 "network service" virtual calls. In order to understand the basic communication that takes place, a simple description of the TP software architecture, related to NCP communication, follows.

The TP 2200/4000 uses a system master architecture, with slave processors acquiring centrally controlled resources from the system master. Each TP contains 4K bytes of non-volatile Read-Only Memory (ROM) code which is used to load the system. This ROM code contains a partial implementation of the X.25 protocol. To load itself, the TP sends a CALL REQUEST packet to the NCP and thereby establishes an X.25 virtual call. The TP's own address and the address of the NCP are encoded in strap bytes within the ROM code, and are easily set for each TP.

From the address of the originator of the CALL REQUEST packet, the NCP determines which software system and which tables are required. Software for both master and slave cards is sent to the TP along with the necessary configuration tables. The TP receives these programs, control is turned over to the master card, and the master card initializes the system. Slave LPUs are then instructed to load their local memory from the main memory. The TP is then operational. The loading process typically takes several minutes, depending upon access line speed and NCP load.

The TP system master initiates network service virtual calls to the NCP. The master informs the NCP any time there is a system malfunction and responds to NCP TDT commands.

For each virtual call established between two DTEs on the network, accounting mechanisms are embedded in the terminating nodes; that is, the nodes which are directly connected to the DTEs. The accounting information for an endpoint of a virtual call is extracted from the local node when a CLEAR CONFIRMATION packet is received from or sent to the DTE. It is formed into a DATA packet and sent to the NCP via a network service virtual call.

5.1.3 NCP Software

Three types of software modules are resident in the NCP:

- Active

  Software modules that collect status reports and accounting data automatically generated by each TP in the network are in the Active modules;

- Operator Activated

  Software modules that generate queries or commands to network TPs to request particular status information or to change the state of network components are within the Operator Activated modules; and,

- Background

  Software packages that do not interact with the network which are used for background functions, such as building network tables and analyzing data that has been collected are in the Background modules.

The NCP software modules are written as application programs. Even those classified as 'Active' must be activated as a system job.

5.1.3.1 Active Modules

The TP Reporting Facility (TPRF) modules and the TP Loader reside in the NCC and are active 24 hours a day. TPRF analyzes, processes, and reports on data transmitted to the NCC by the network TPs. The TP Loader handles software and table load requests from the TPs.

TP Reporting Facility

The network TPs dynamically report to the NCP whenever a significant change in status occurs. Alarm messages are sent to the reporting program and are displayed on the NCC console. Each message identifies the network address of the TP and is time and date stamped by the NCP.

The alarm messages sent to the NCP report the following types of events:

- the status of LPU and CPU cards
- "coming up" messages
- memory errors
- trunk line status
- X.25 LAP handler status
- X.25 LAPB handler status
- terminal handler status
- host handler status
- correctable software errors
- system status as a whole (eg. CPU interrupts, etc.)
- TP power supply status

5.1.4 Operator Activated Modules

The Telenet Diagnostic Tool (TDT) is an on-line aid that permits the NCC operator to examine and change the state of network components. It is used to assist in diagnosis and, in many cases, to correct a network problem. A full description of the commands available to the private network customer is contained in Appendix A, Telenet Diagnostic Tool.

5.1.5 Background Modules

Network configuration tables for the TPs are built, updated and compiled into a load module using application programs written in PRIME FORTRAN. They are used to build files that are loaded into the network via operator activated modules.

Similarly, application programs may be written to perform analysis of data collected by TP Reporting Facility. These programs are typically user-written and run on a demand basis when a particular aspect of network operation requires careful study.

The UPDATE, PORT and SPORT programs are used to build network configuration tables and are detailed in the Telenet "UPDATE," "SPORT," and "PORT" manuals. These programs, and a program for creating user IDs and passwords (see Section 6.4), are provided only to private network customers having their own NCC.

5.1.5.1 UPDATE

UPDATE is a data base editor designed by Telenet to update the NCC data base upon which the network's master configuration and control tables are built. UPDATE permits the user to add new DTEs or hosts to the configuration tables and/or to change existing configurations. The configuration tables are subsequently loaded into the network under operator control or upon automatic request for a reload from a TP 2200/4000.

5.1.5.2 PORT and SPORT

PORT is an interactive program that builds or modifies parameter files for ITI PAD interface ports (both host and terminal). The user is prompted for answers to questions that cover all possible parameters.

SPORT performs in a similar fashion for BMTI PAD interface ports.

5.2 CUSTOMER NETWORK CONTROL CONSOLE

The Customer Network Control Console (CNCC) performs a subset of those network management functions provided by an NCC, in particular the event reporting portion of TPRF and TDT access. The CNCC user time-shares the Telenet NCPs.

Accounting

When a virtual call is established, a unique connection identifier is assigned. This identifier is identical for both ends of the call. When a call is cleared, both ends of the call send an accounting record, which contains the unique identifier as part of the data, to the NCP. The accounting system uses the identifier to pair "half-call" records to produce a "full-call" record.

Call records currently contain the following information:

- source DTE address
- destination DTE address
- unique connection identifier
- call clearing time stamp
- call duration
- traffic volume for the virtual call
- traffic volume sent to the source DTE
- traffic volume received by the source DTE

Traffic volume is measured in terms of "accounting volume units." The accounting volume unit is set as a subscription time option and may be set as 1, 2, 4, 8, 16, 32, 64, or 128 bytes. The network default is 128 bytes.

When the NCP receives accounting information, it is immediately written onto a disk file. The disk file is transferred to tape under manual control by a background program.

TP Load

TP Load is a program with dual areas of activity. It serves as the primary vehicle for both detailed trouble reporting and loading activities at the NCC.

As a trouble reporter, TP Load's primary purpose is to explicitly display, and provide storage for, the data associated with any given node failure. The program also provides network nodes with a recording facility for the delineation of software problems.

In the role of "NCC Loader," TP Load essentially attends to node table/software loading.

Messages associated with both elements of the TP Load program are displayed continuously, as events occur.

The TDT commands available to the CNCC are restricted by an account/password feature. A restricted set of network addresses may be examined or have their status modified by the CNCC operator. A detailed discussion of the TDT commands and their application is contained in Appendix A of this document.

A CNCC has, at a minimum, one status console to which all events that are reported by TPs are forwarded. In addition, all event reports are stored in a message file and are optionally available, in magnetic tape format, for off-line processing.

The CNCC can be defined with more than one status console. Status messages can be routed to selected consoles based on message type and source address. This facility enables the CNCC user to dedicate different status consoles for different classes of alarm messages or for alarm messages from different portions of the network. The manner in which this "selective filtering" of alarm messages is implemented provides the CNCC customer with the ability to restrict the volume of messages received on the status console. In addition, it furnishes the ability to explicitly capture selected types of alarm messages on specific status consoles for immediate operator action.

It should also be noted that the status consoles will periodically print out a time stamp to indicate that the system is running correctly, even if there are no events to report.

5.3 ACCOUNTING

The third management tool available to a private network customer is the statistical and analytical data gathered by the NCC during on-line operation.

Accounting information is gathered from the TP and is made available on magnetic tape to the CNCC customer. The accounting tape provided to the private network customer is divided into 3 basic record formats: Header Records, Detail Records, and Trailer Records.

Header Records

The Header Record contains customer identification information as to whom the tape is being generated for, and the time frame for which the tape applies.

Detail Records

The Detail Records contain specific details for each call, such as connection origin, access type, speed, destination, traffic volume, call duration, and traffic and connect charges.

Trailer Records

The Trailer Records contain totals for connections, time, traffic volume, and charges for traffic and connections.

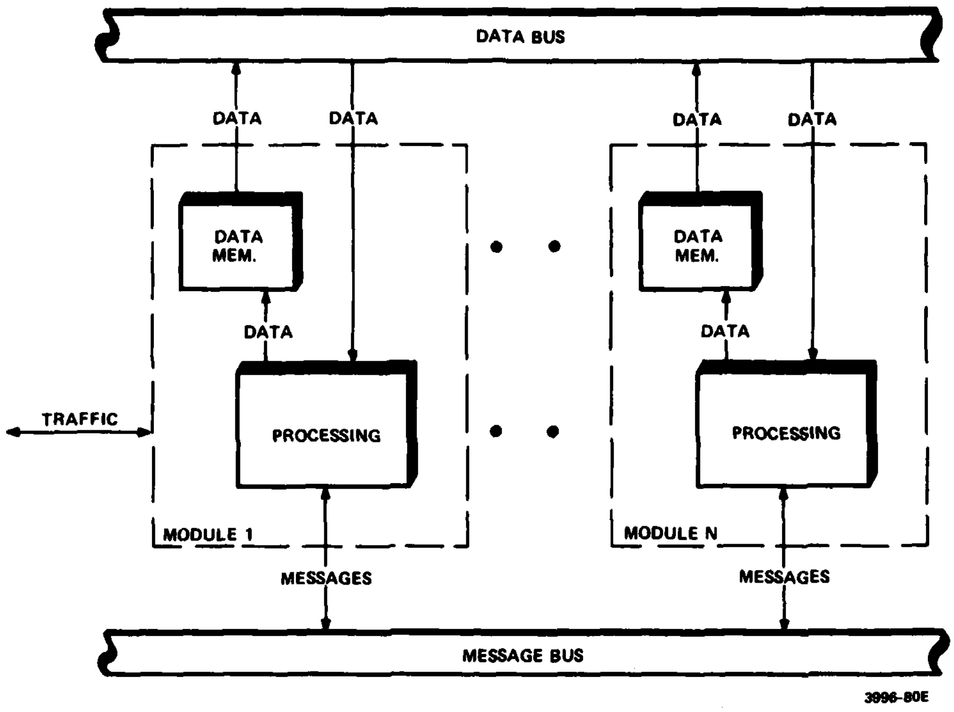
APPENDIX C

DISTRIBUTED TEST BED ARCHITECTURE

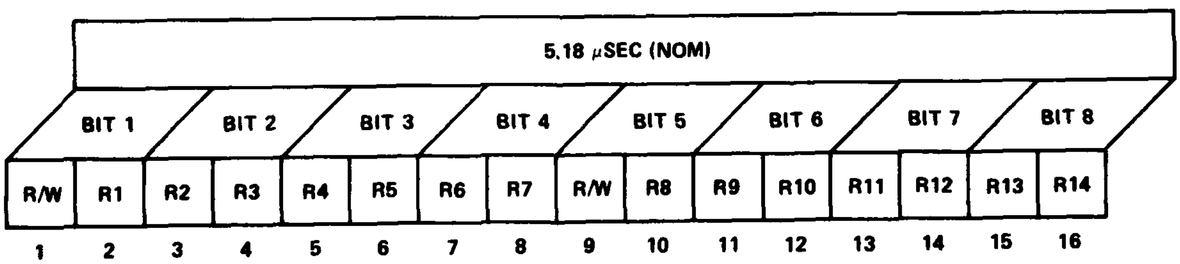
The specific distributed architecture chosen to implement the test bed meets the appropriate timing constraints and also exhibits flexibility to implement several switching concepts and provides ease of expansion. In this section, the basic switch architecture will be described, including the modules that compose the switch, the timing scheme, data bus control and access and trunk interface.

Figure C-1a shows a top level view of the basic architecture. The architecture is built around a data bus and a message bus, with up to 14 modules connected to ports on the data bus. Each module consists of input and output processing elements (a blend of hardware/firmware/software), input data memory, and memory multiplexing, address decoder and enable elements. Data throughput flows from the input stream through the processing element of the imput module into the data memory. Then it flows out of data memory, across the data bus, through the processing element of the output module and into the outgoing data stream.

Data memory access schemes consist of local read/write access or global read-only access. The timing to make this happen is shown in Figure C-1b. Here is a basic time frame of 5.18 usec., based on the cycle time needed to accumulate 8 bits transmitted at a T1 rate. Each frame is subdivided into 16 slots to be used as memory access slots of 320 nanoseconds. All writing is done by an input processing element to its own memory twice each cycle (i.e., during slots 1 and 9). One slot is used to store input data; the other is used to write other data (e.g., header, CCS) into data memory. During these times, there is no data bus activity because each input processing element is acting on its memory. This leaves 14 time slots to be shared by the output processing elements for the global read-only accessing of other data memory. During this time, each module on the bus is assigned a read slot to gain control of the address and data lines. Thus, during slot 2, module 1 can read from any data memory; during slot 8, module 7 can read from any data memory, etc. This arrangement allows up to 14 separate output or reading modules to access memory at the T1 rate on a nonblocking basis. A separate message bus helps make this possible.



(a) Basic Architecture



R/W = LOCAL READ/WRITE SLOT FOR ALL MODULES

- EACH MODULE WRITES DIRECTLY INTO (OR READS FROM) ITS OWN DATA MEMORY

Rn = GLOBAL READ SLOT FOR MODULE n

- MODULE n GAINS CONTROL OF DATA BUS ADDRESS LINES TO READ FROM ANY DATA MEMORY

3996-80E

(b) Data Bus Timing

Figure C-1. Basic Test Bed Architecture

Figure C-2 shows the overall block diagram for implementation of the system. The idea of distributed processor control with microprocessors controlling front end (external interface) and output sections, according to their own unique protocol, and then using a common protocol for internal switch functions and intermodule switching, is used extensively. Since this is a test bed whose prime task is to evaluate switching algorithms, not hardware efficiency or cost, extensive redundancy to provide graceful degradation in the event of failure is not greatly emphasized.

A distributed function system has the data transfer and the message transfer taking place independently (see Figure C-3). Each module of the data bus is assigned a time slot (See Figure C-1b) to read from the combined/distributed memory. The bus is run under the control of the Data Transfer Timing that acts as a synchronous multiplexer controller. The message bus is a synchronously controlled bus that allows control data, such as location of packets in memory, to move between the different modules. This bus is controlled on a rotation scan basis with safeguards added to prevent bus hogging.

This dual bus system allows the microprocessors to exchange control information on one bus and switched data on the other. In the case of packet switching, for example, packets are received by front end modules and stored automatically in the main data memory. The main processor is then given via the message bus, the starting address and byte count of the packet stored in the main data memory. It looks up by way of the data bus, the message type (e.g., PAR, PVC, controL, CCS, etc.) and identifies the routing and destination address in order to determine the corresponding module which will use the packet. The main processor then uses the message bus to inform the output module of the starting address and byte count. Any internodal communication such as CCS and NMC information are stored in a separate section of data memory directly by the processors (local read/write cycle) with the appropriate link level information; however, they are handled just the same as if they were packet data.

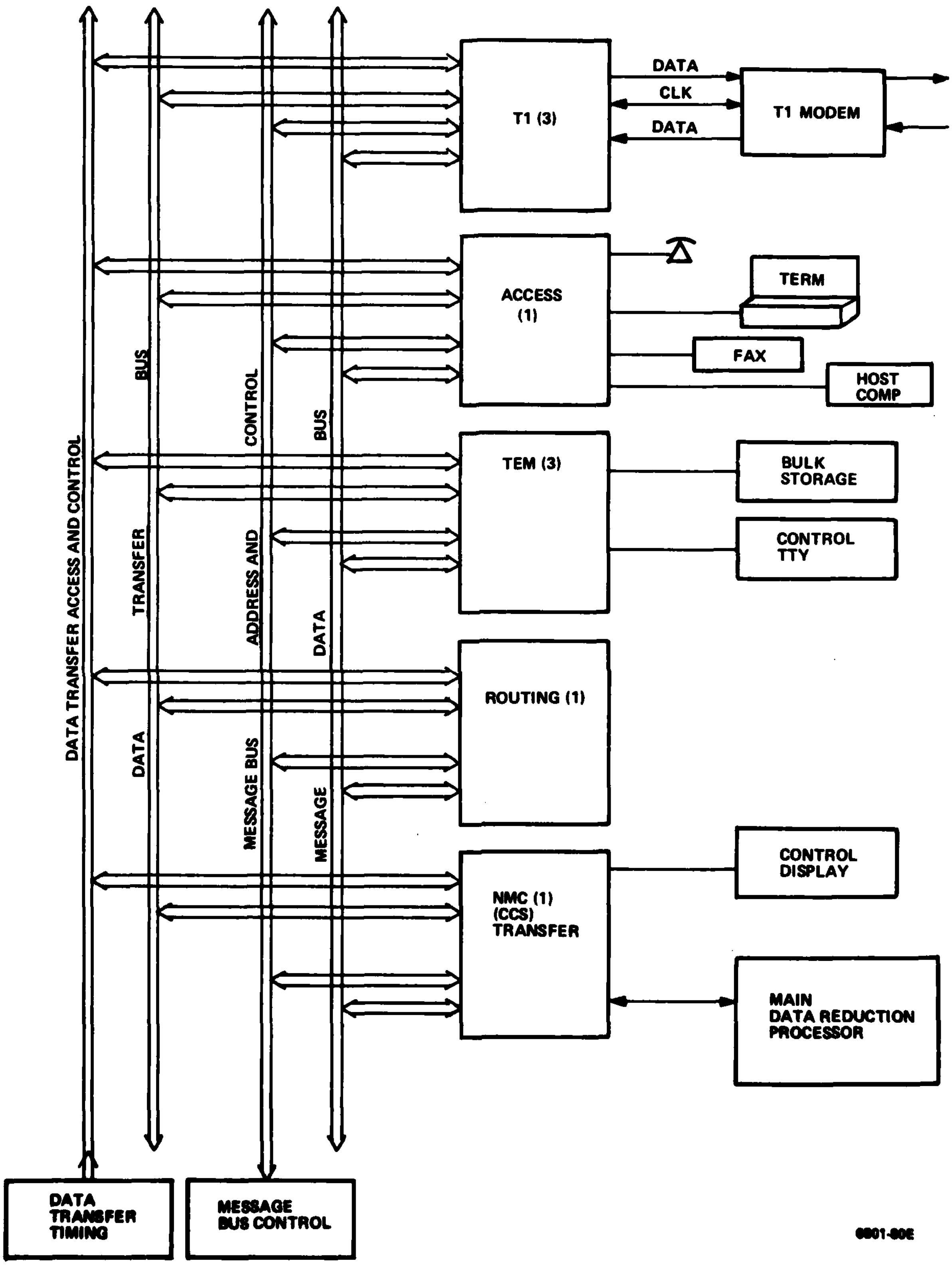


Figure C-2. Block Diagram of Distributed Test Bed Architecture

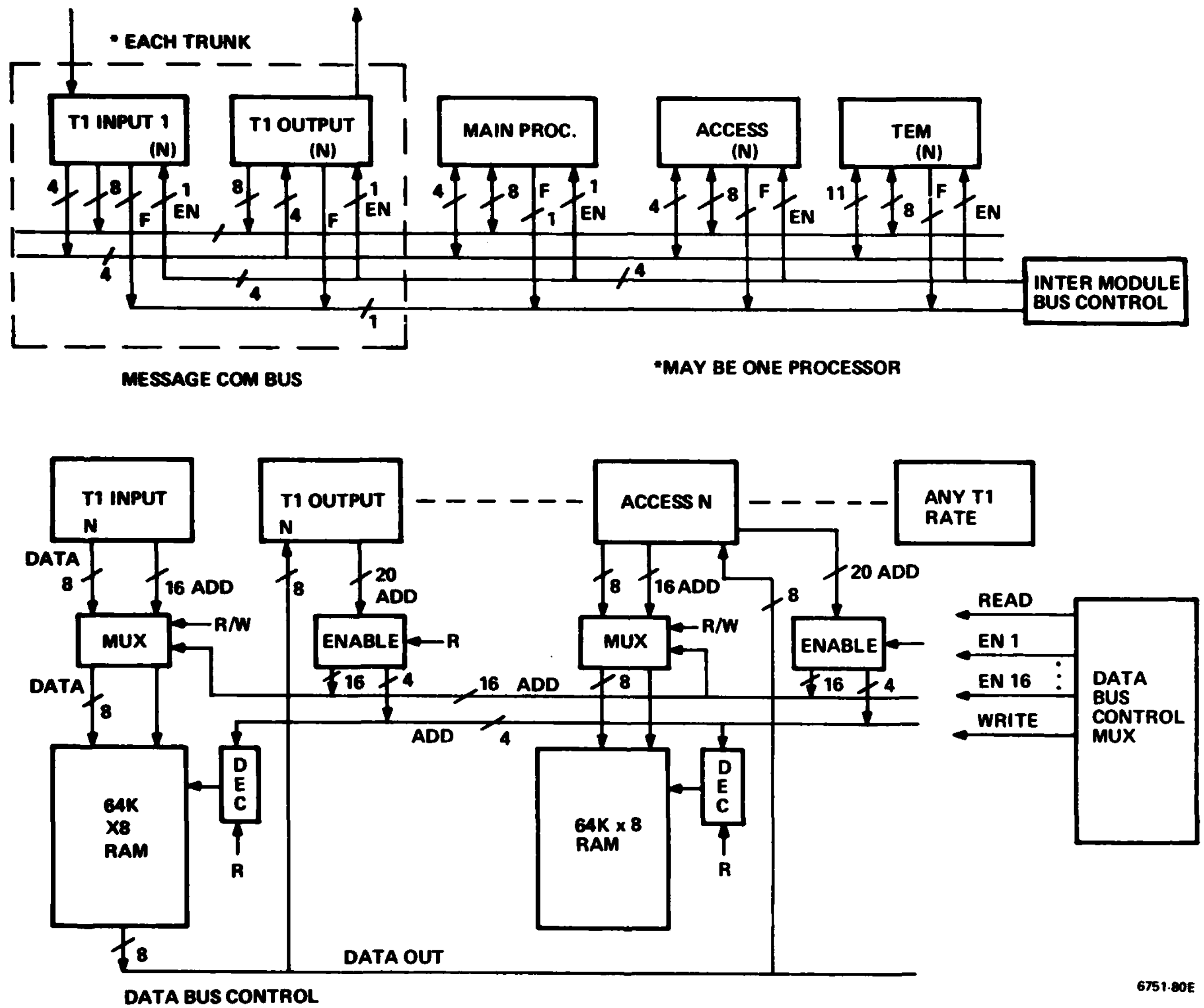


Figure C-3. Message and Data Bus Control

The data bus control is a hardwired clock sequencer (multiplexer) that allows all writes to take place in given time slots; the rest of the time slots in the 5.18 usec byte time are used for read access. Figure C2 is an example of a system that services 3 T1 trunks, 1 access area at an equivalent T1 rate, a main processor, and 3 TEM's.

An arrangement using 16 time slots requires a memory with an access time less than 320 nanoseconds. This is readily achievable using present day technology. In the baseline design for the test bed, the data transfer for these 14 modules is done on an 8 bit wide basis, thus producing a capable data I/O rate of over 21 MBS. The message bus is also 8 bit wide parallel transfers but rather than having a byte by byte interleaving of data like the data bus, an entire message is transferred by sequential transfers. The message bus arbitrator polls the message ready bits of each access function to determine bus requests. Based on sequence of arrival and priority, the message bus arbitrator will grant bus control to one of the requesting modules. The module will then transfer the message data by presenting destination address and information.

After each single message transfer, the bus is surrendered by the originating unit and a new request and grant cycle must be completed before a second transfer can take place. The bus arbitrator will cycle through other requests by some predetermined algorithm using a combination of last served, first in, and priority to determine the next module to receive service. The arbitrator also has timing protection to prevent bus hogging by any of the modules.

Each module interfacing to the message and data buses will use a standard interface technique (See Figure C3). The message bus interface uses FIFO's so that full asynchronous and independent operation can go on. The interface processor has two FIFO's for the bus interface. One is used for outgoing messages and one for incoming messages. In normal operation, the processor has messages or information to be transferred to or from other modules. When information (e.g., packet data location in memory) is to be transferred, it is loaded into the FIFO along with the destination address, and a flag is raised. The processor then continues to perform other

chores. The message bus arbitrator recognizes the message ready flag and effects the transfer. Data is loaded into the receiving FIFO and a flag indicating the presence of a message is raised. The receiving processor can then service the message at its leisure. The data bus interface is split into a transmitting (output) and receiving (input) function. Because of the variable bandwidth nature of the bus there must be buffering within the switch even for the circuit switched traffic. This is due to asynchronous framing of the trunks and the relative slip nature of the more advanced switching concepts (including TASI).

As can be seen in Figure C-4, there exists a three frame storage for the circuit region. This is necessary to cover all possible synchronizing cases of relative frame start, variable frame effects, and a relative slot assignment within the circuit switched region. The delay for these cases range from a small fraction of a frame to almost three full frames for each switch. Average delay is 1.5 frames. Delay remains relatively fixed for a call, however, since it can vary only one frame length per switch, with only the output switch jitter having an effect on the call jitter. This can be overcome by the appropriate buffering in the output queue. The analysis for allowable circuit region size and memory storage requirements is based on a hardware controlled data transfer. This assumption is required to make the set-up time for data transfer independent of processor loading and traffic.

The synchronizing for data transfer between input and output modules for circuit control is done under the frame control reference counter. Every input module has the same starting addresses for its three circuit region storage areas. The starting addresses are stored in the sequence of 0, 4K, 8K for counter numbers 0, 1, 2. In this way, the outputs are always reading out a 3 frame delay in the circuit area as shown in Figure C4. This basic step takes care of the continuing call case and the fixed delay that guarantees the data is available from the inputs when needed. The output map control is a dual function to that of the input map control. Each output is notified of the index and byte count of the calls that are routed to

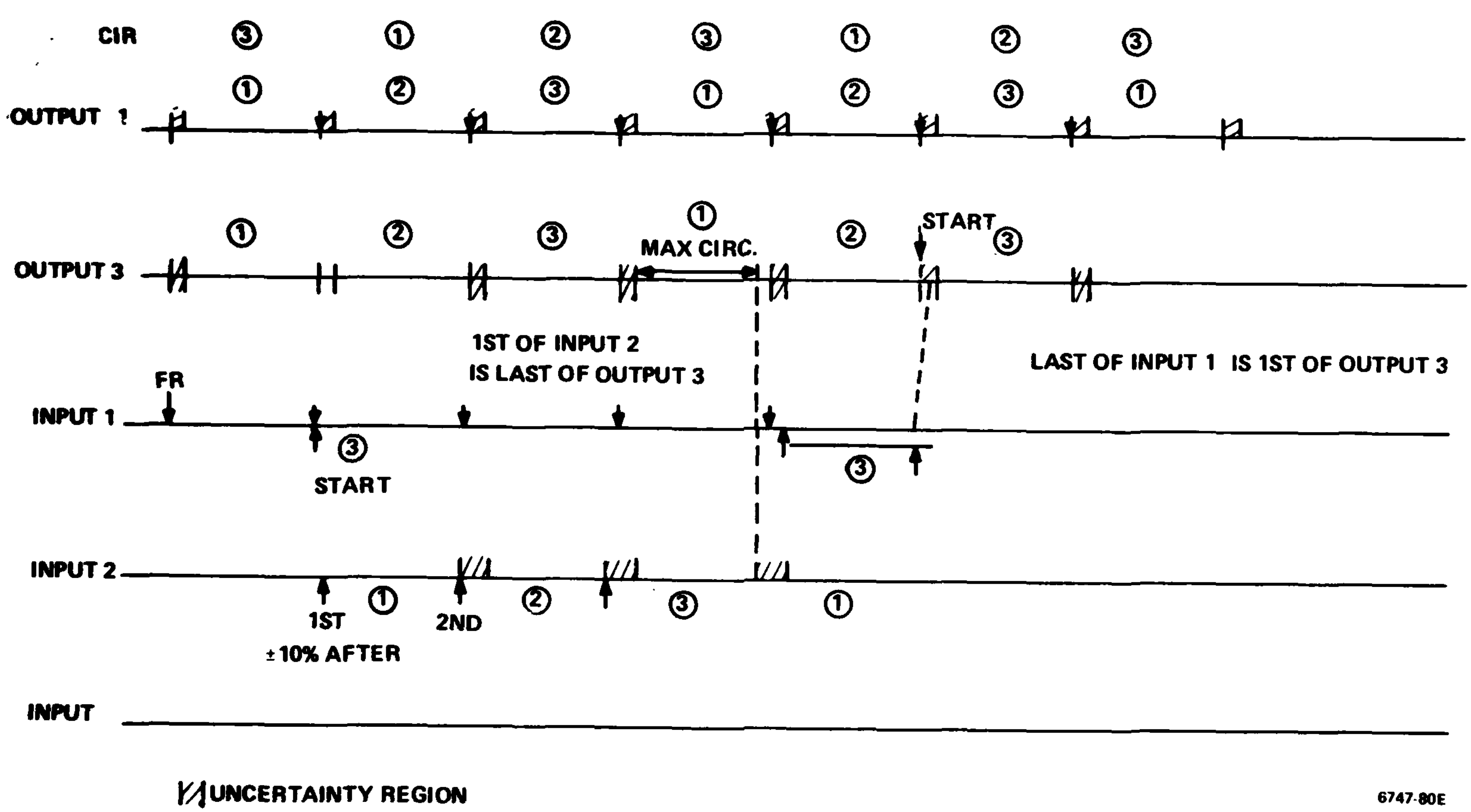


Figure C-4. Variable Boundary Frame Effects on Storage Requirements

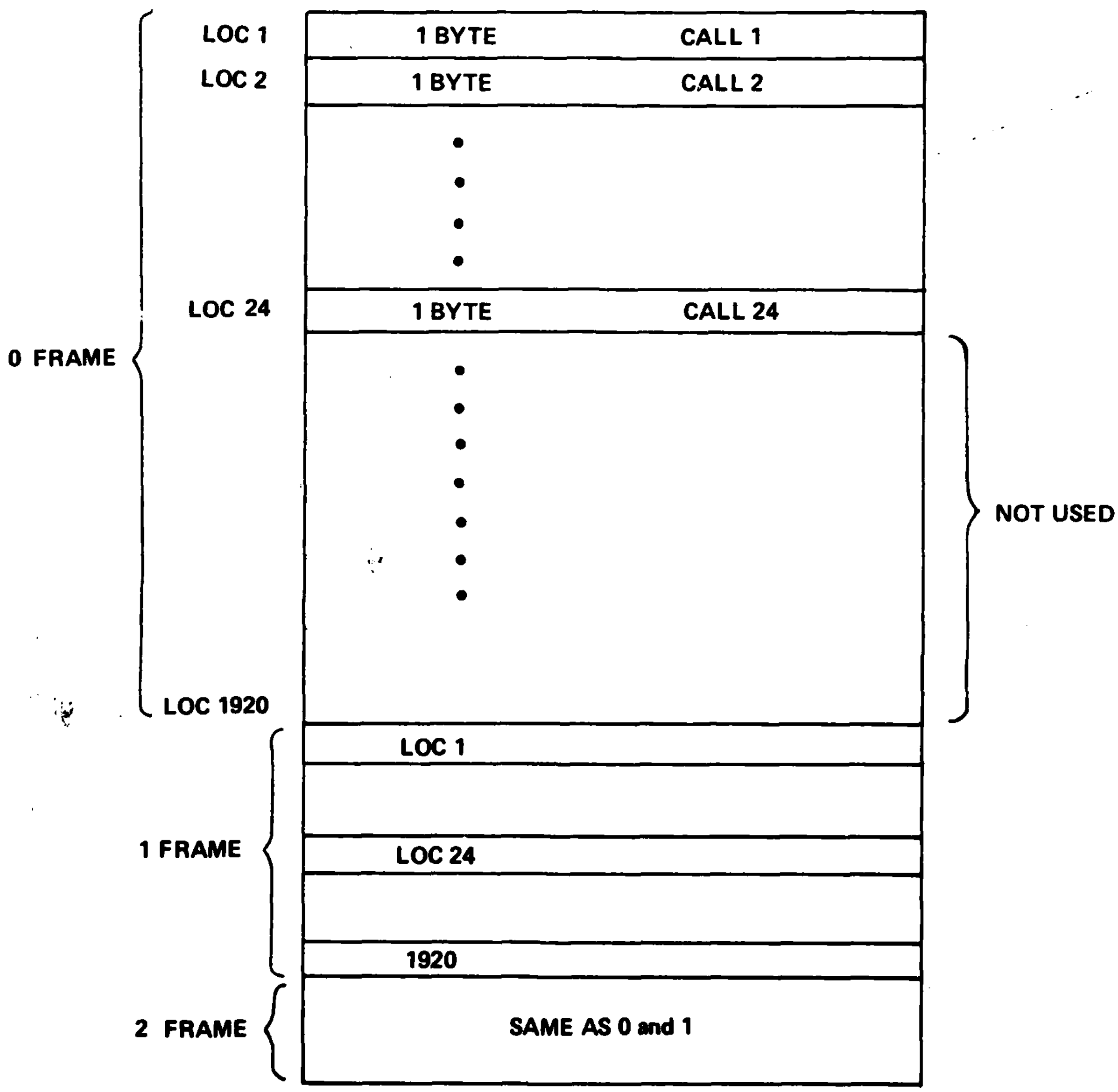
that output. With the information from each of the inputs, including access area functions, it builds its own output map (Figure C-5) containing identity of the call data by the input module, the relative memory location (index) and the byte count or bandwidth of the call. As changes in the connect requirements come through the output module builds a new map.

In the case of a standard 64K bps PCM circuit switch, each frame is 125 usec long and contains 1 byte of information per channel (See Figure C-6). For the advanced hybrid trunks with variable frame rates, a much longer multiplexing map is required (e.g., a 10 ms. frame requires 1920 entries for full flexibility). Similarly, the data storage area of all input modules must accommodate a three frame storage capacity. This leads to a 6K storage requirement (3 frames x 1920 bytes per frame) for circuit switched data in a hybrid system.

Since all data transfer is by way of the shared memory access, a similar mechanism is used for packet data transfer (Figure C-7). An output module receives information about the location, priority, etc. of the packet to be transmitted via the message bus. When it is time to input the packet, the data is transferred on a byte-by-byte basis using a DMA-like process where sequential locations of stored data are addressed and data is read on each bus cycle. External interfaces (packet, circuit or hybrid) are customized to the standard system data bus which requires that the information for 3 frames of circuit switched data be stored in input memory. Also required is enough storage for all packet data so that normal operation will not cause a memory overflow.

For the test bed, the data bus and each access interface must be designated to accomodate a 3 frame interval ranging from 125 usec to 10 msec.

Access modules, and in general all interface modules, use a standard format for bus interface. The message bus will handle high speed data transfer, controlled by the bus arbitrator, and will transfer one message at a time. The data bus interface will be in the form of a general memory addressable scheme with the output module presenting the address of the data desired, under timing of the data



(a) INPUT FRAME MEMORY FOR CIRCUIT-SWITCHED DATA (FIXED BW MAP)

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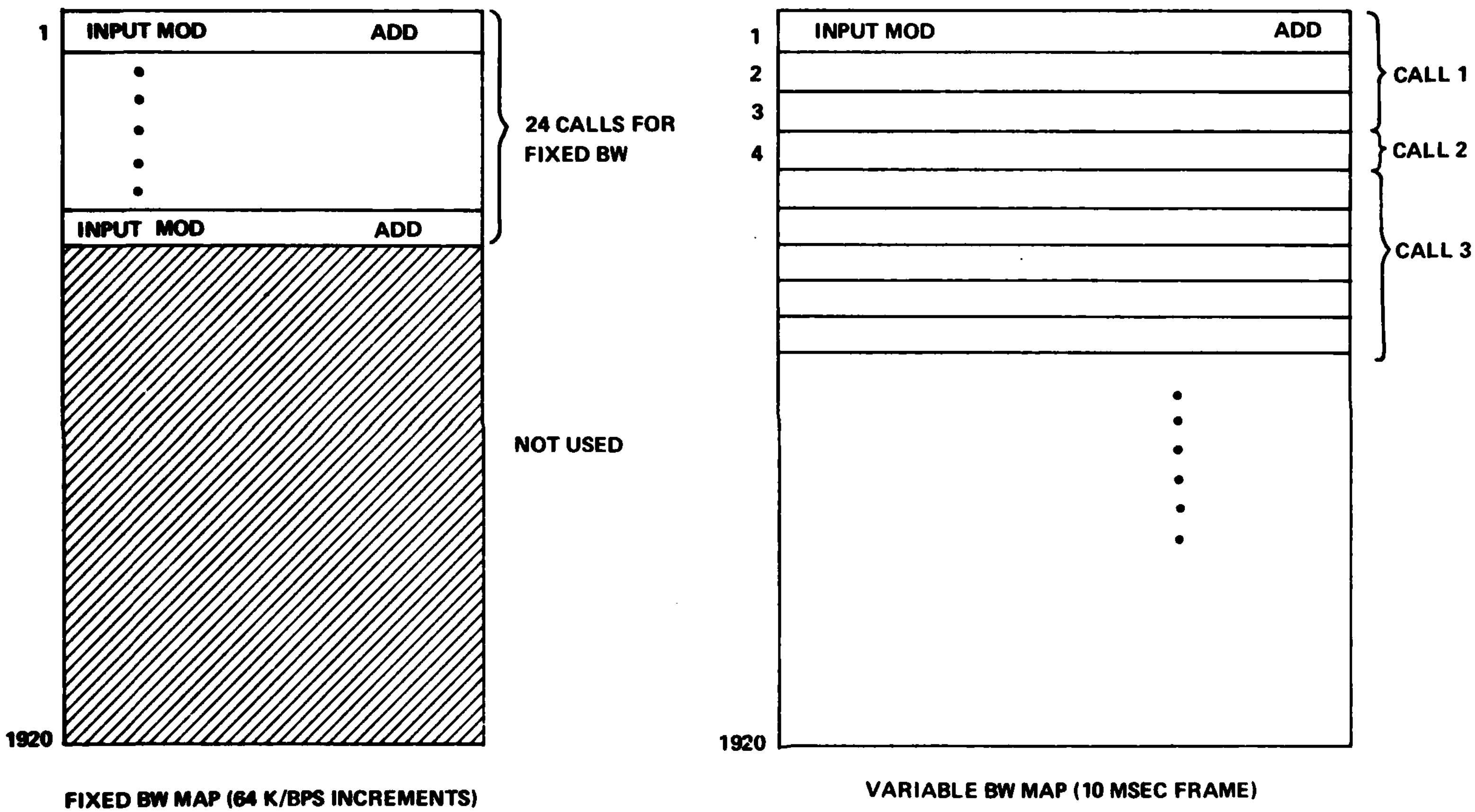
Figure C-5. Construction of the Output Circuit Switched Map



(C-5b) OUTPUT CIRCUIT — SWITCHED MAP

Figure C-5. Construction of the Output Circuit - Switched Map (Continued)

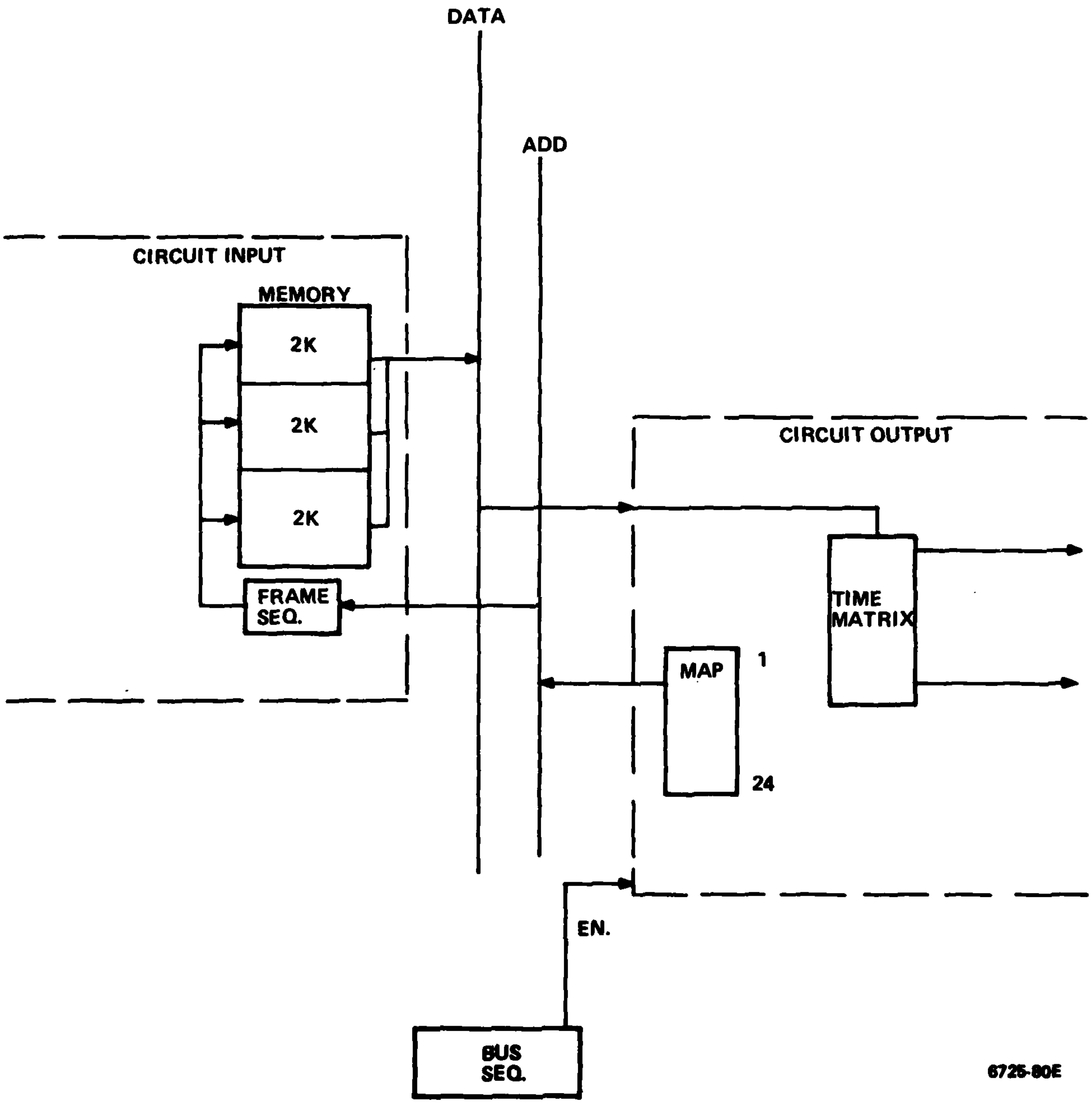


Figure C-6. Circuit - Switched Data Transfer

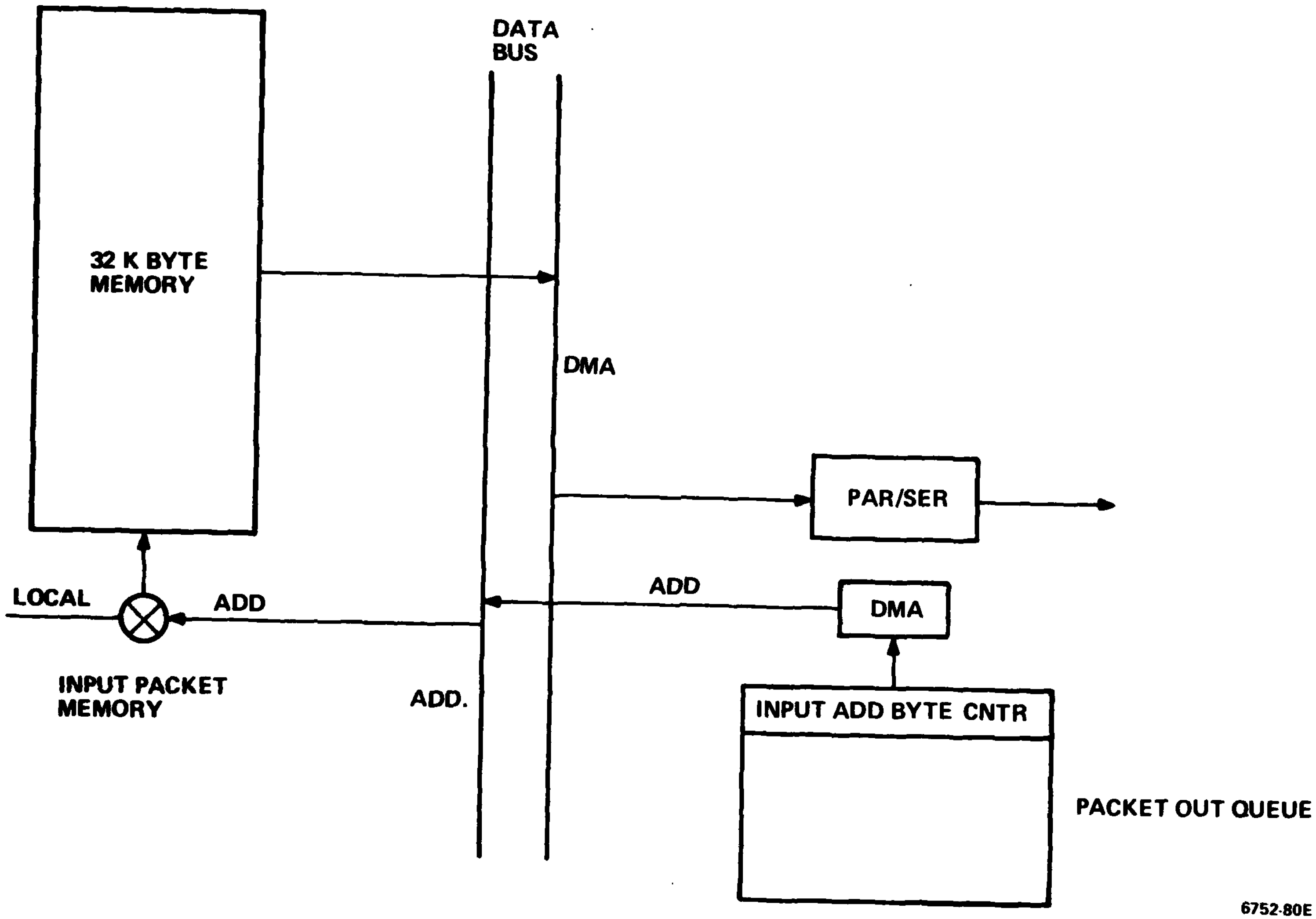
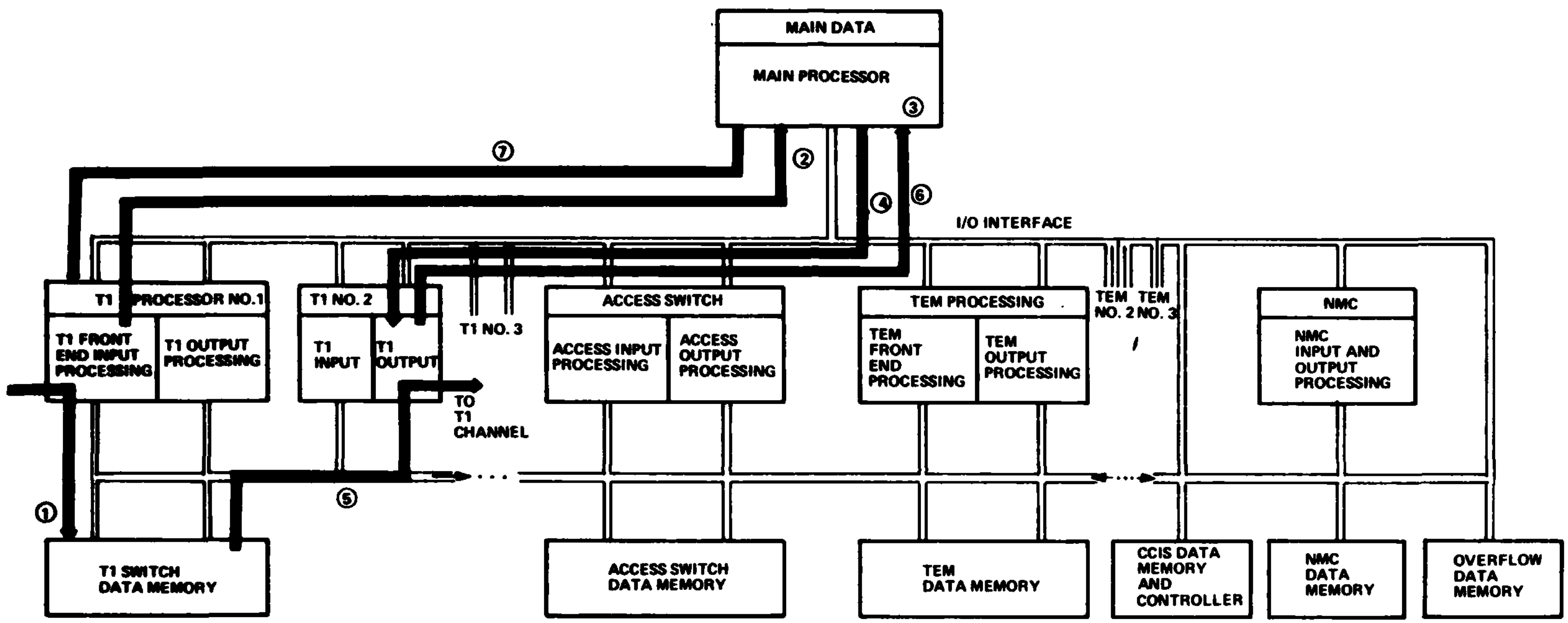


Figure C-7. Packet Input Memory & Output Map

bus control. Information defining the location of circuit switched calls and the starting location and byte count of packet information is sent by way of the message bus.

A software data flow for a typical transaction is shown in Figure C-8. This example is for transfer of a packet on a hybrid trunk. Although an advanced application, the same principal would also apply for a circuit switched call of a standard packet switched data transfer.

In the baseline switch the circuit switch access area serves as a concentrator. Access processing functions control the multiplexing to the bus interfaces and provides telephone interface protocol such as digit collection. The access area will be designed for standard 64K bps Channels - with this assumption, input memory and the output map functions are simplified due to the fixed nature of the call bandwidth. The input buffer will be designed to accomodate a 10 msec frame but will use only a 125 μsec format for the early experiments. Routing, resource allocation and other non-interface functions will be performed by the central processor. In later experiments, this processor will handle similar functions for both the packet and hybrid configurations.



PACKET SWITCHING SCENARIO

1. FRONT END HARDWARE DETECTS FLAG AND STORES PACKET INTO DATA MEMORY ACCORDING TO SIZE.
2. T1 INPUT PROCESSOR NO. 1 SENDS FOLLOWING INFORMATION TO MAIN PROCESSOR:
   a. FIRST 3 TO 4 WORDS OF PACKET (HEADER INFORMATION),
   b. STARTING ADDRESS OF WHERE PACKET IS STORED IN T1 DATA MEMORY.
3. MAIN PROCESSOR TRANSLATES HEADER AND DETERMINES THAT PACKET IS TO BE OUTPUT ON T1 OUTPUT MODULE NO. 2.
4. MAIN PROCESSOR GIVES ADDRESS OF DATA TO T1 OUTPUT PROCESSOR NO. 2.
5. DATA IS SENT OUT ON T1 OUTPUT MODULE NO. 2.
6. T1 OUTPUT PROCESSOR NO. 2 INFORMS MAIN PROCESSOR THAT PACKET HAS BEEN SENT.
7. MAIN PROCESSOR INFORMS T1 INPUT PROCESSOR NO. 1 THAT PACKET HAS BEEN SENT. T1 INPUT PROCESSOR NO. 2 FREES UP THE CORRESPONDING MEMORY.

Figure C-8. Incoming T1 to Outgoing T1 Packet Transfer

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